VOIP GATEWAY

SIP

- 2PORTS
- 4PORTS
- 8PORTS

DUAL NETWORK OPERATION MANUAL

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Operation Manual

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1. Introduction

Product Overview

The stand-alone VoIP Gateway carries both voice and facsimile over the IP network. It supports SIP industry standard call control protocol to be compatible with free registration services or VoIP service providers' systems. It works in two different modes: UA (User Agent) or Server. As a standard user agent, it is compatible to all well-known Soft Switches and SIP proxy servers. While running the optional server software, the gateway can be configured to establish a private VoIP network over the Internet without a 3rd party SIP Proxy Server.

There are 3 types of gateways in the same series: 2 ports, 4 ports and 8 ports (voice ports, FXS and/or FXO). The gateway can be seamlessly integrated to existing network by connecting to a phone set, PBX, key telephone system, fax machine or PSTN line. With only a broadband connection such as ADSL bridge/router, Cable Modem or leased line router, it allows you to gain access to voice and fax services over the IP in order to reduce the cost of international and long distance calls.

In addition, the in-built 4 ports Ethernet switch supports comprehensive Internet gateway functions to accommodate other PCs or IP devices to share the same broadband stream. QoS function allows voice and data traffic to flow through where voice traffic is transmitted in the highest priority. With TOS bit enabled, it guarantees voice packets to have first priority to pass through a TOS enabled router.

With the support of DDNS, it makes the gateway reachable by its domain name where the IP address is dynamically assigned by the ISP. It helps users to host a web site or mail server in a PPPoE or DHCP network. By enabling the CDR function & setting up a simple server, administrators are allowed to log in and view all call records such as call duration, time and date of calls and latency.

The gateway can be assigned with a fixed IP address or by DHCP, PPPoE. It adopts the G.711, G.726, G.729A or G.723.1 voice compression format to save the network bandwidth while providing real-time and toll quality voice. In addition, in the event that the power supply fails or Internet connection is lost, the gateway can automatically divert the FXS end to the PSTN network on the FXO port so users can still use the conventional PSTN line to make calls. This feature is especially useful while dialing emergency calls (ie. 911).

Product Features

- SIP (RFC 3261) compliant
- Supports 2 / 4 / 8 simultaneous FAX / Voice calls (port number differs between models)
- 4 Ethernet switch ports with IP sharing functions
- Optional server enables small businesses to build up private VoIP network (SIP model)
- QoS support guarantees voice bandwidth in a busy network
- Supports IP TOS (Type Of Service)
- Internet gateway functions: DHCP server, NAT, Virtual Server, DMZ, IP/PORT/MAC filtering
- T.30 (G III) / Real Time T.38 / Secured T.38 Fax Relay
- Feasible for Fixed IP address or dynamic IP address network (PPPoE / DHCP client, support DDNS)
- Configurable Hot Line feature
- Supports IP-to-PSTN / PSTN-to-IP applications
- Life-Line support (IP / power failure over relay)
- NAT traversal - STUN and UPnP (optional)
- Pass through NAT
- Call Detailed Records (CDR)
- Web-based firmware upgrade
- Caller ID Delivery
- Easy Configuration by IVR and Web-based GUI
- Greeting message
- Echo Cancellation: G.168 compliant
- Voice Activity Detection (VAD) & Comfort Noise Generation (CNG)
- Dialing: DTMF, PULSE (optional)
- Signaling Protocol: Loop Start
- Adaptive Jitter Buffer and Programmable Gain Control
- Internet Gateway Functions : DHCP server, NAT, Virtual Server, DMZ, IP/PORT/MAC filtering

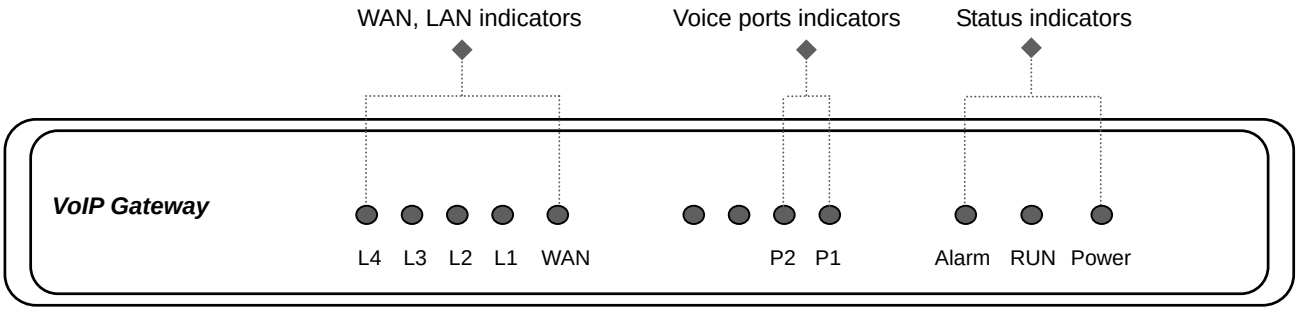
CALL features (optional)

- Call hold
- Call waiting
- Call forward
  - Unconditional (follow me)
  - Busy forward
  - No answer forward
- Call transfer

Hardware Description

2 ports gateway model: 2S / 2O / 1S1O

Front Panel



- Power Indicator: Green light indicates a normal power supply.
- Run Indicator: Blinking green light indicates normal operation.
- Alarm Indicator: When the system starts up, the red light will blink. It also indicates the gateway's abnormal operation.
- P1 – P2 stands for Port 1 – Port 2 (Line 1 – Line 2)
- WAN stands for the WAN Port Indicator.
- L1 – L4 stands for the LAN Port Indicator.

✓ **When starting up the system, the Alarm, Run, and Power indicators will light up. After about 40 seconds, the Alarm indicator will go off, the Run indicator will blink in green, and the Power indicator will stay green under normal operational conditions. If the Alarm indicator continues to blink, it means the system is currently communicating with ISP and has yet to obtain an IP address.**

✓ **When the WAN is connected, the WAN indicator will light up in green and if data is being transmitted over the Internet, the indicator blinks in green and orange.**

Rear Panel

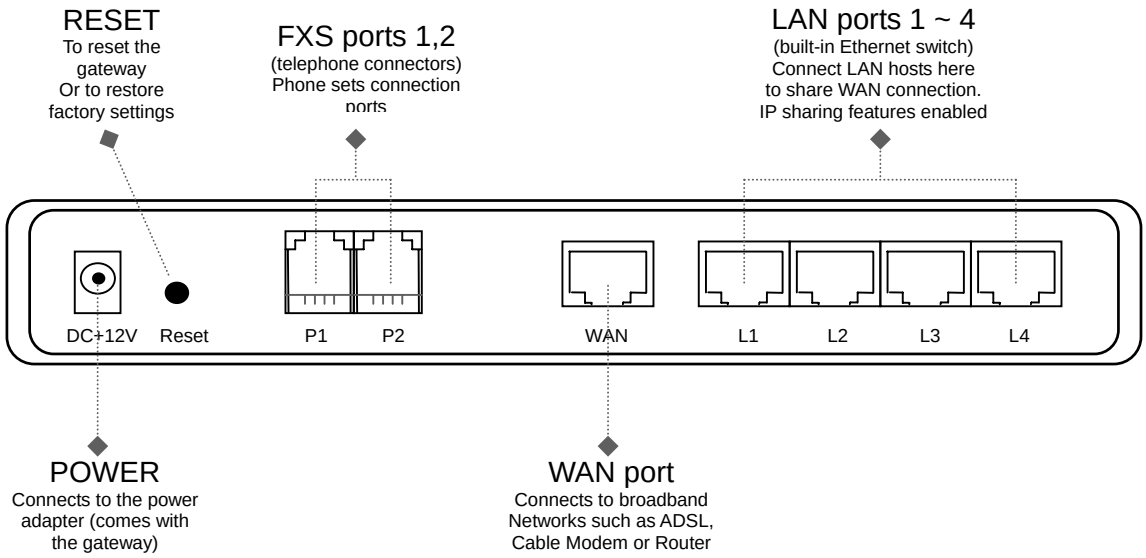
2S **:** P1-P2 are all telephone ports (FXS) to be connected to a telephone set or PBX co ports (PSTN ports).
2O **:** P1-P2 are line ports (FXO) to be connected to a PSTN (telephone lines from a PSTN.)
1S1O **:** P1 is a telephone port (FXS), and P2 is a line port (FXO). Before the power is connected or in the occasion of a power failure, P1 will be relayed to P2 for emergency calls.

NOTE: Do not connect FXS ports to each other. Also, do not connect any FXS port directly to a PSTN line or internal PBX. If any of these are done, your 4S may be damaged.

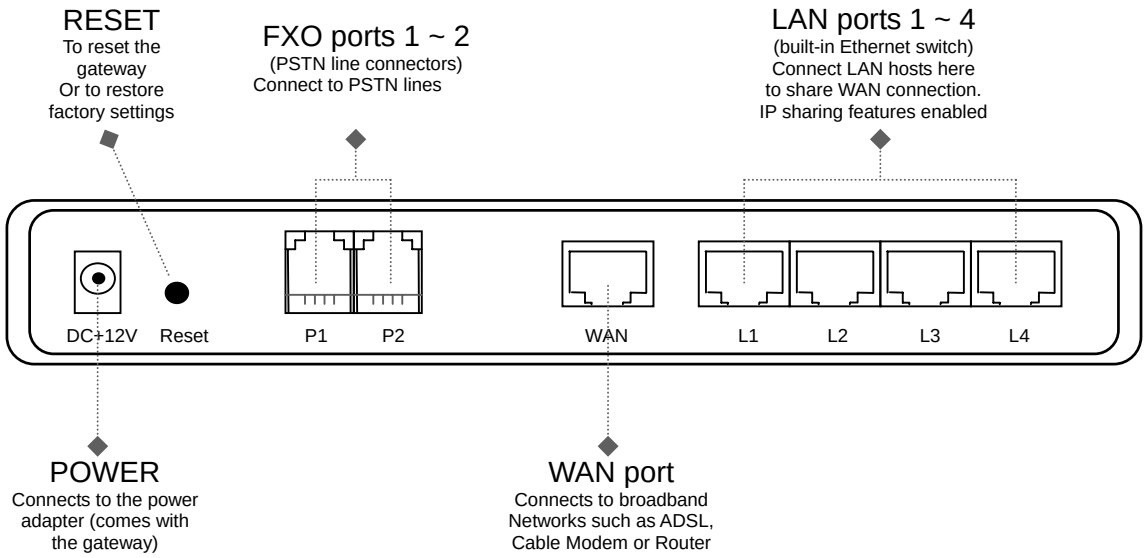
Please use the power adapter comes with VoIP gateway. If using adapters other than the one comes with VoIP gateway, it may cause problems and will affect the warranty of the product.

Restore to factory default: (IP address, User's. Name and Password)
(1) Pull off the power plug.
(2) Press reset (do not let go of the reset button).
(3) Plug the plug back into the socket (do not let go of the reset button).
(4) Let go of the reset button after 6 seconds. Factory settings will be restored.

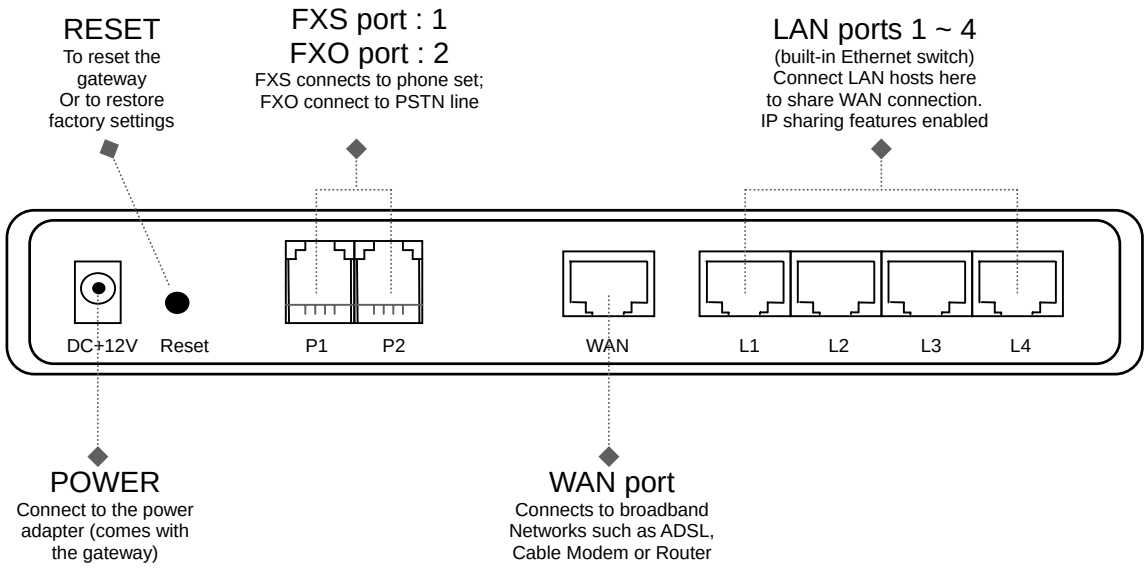
2S Model (2 FXS ports)



2O Model (2 FXO ports)

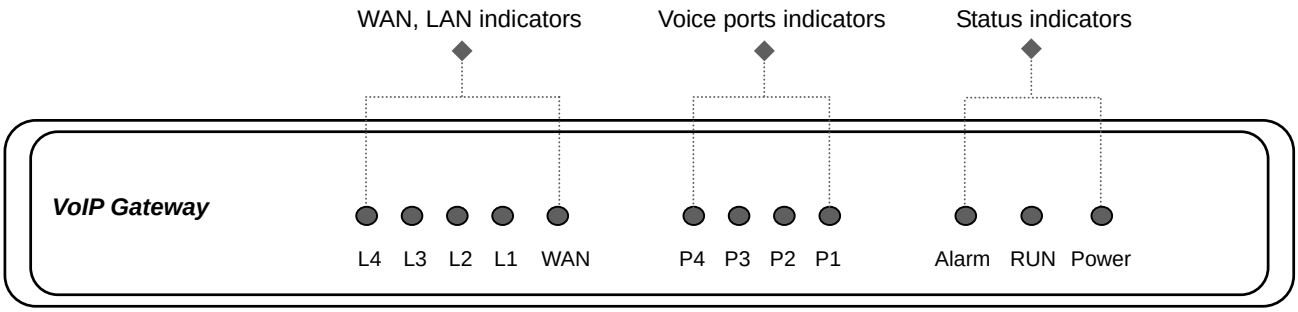


1S1O Model (1 FXO and 1 FXS ports)



4 ports gateways model: 4S / 4O / 2S2O / 3S1O

Front Panel



- Power Indicator: Green light indicates a normal power supply.
- Run Indicator: Blinking green light indicates normal operation.
- Alarm Indicator: When the system starts up, the red light will blink. It also indicates the gateway's abnormal operation.
- P1 – P4 stands for Port 1 – Port 4 (Line 1 – Line 4)
- WAN stands for the WAN Port Indicator.
- L1 – L4 stands for the LAN Port Indicator.

✓ **When starting up the system, the Alarm, Run, and Power indicators will light up. After about 15 seconds, the Alarm indicator will go off, the Run indicator will blink in green, and the Power indicator will stay green under operational conditions. If the Alarm indicator continues to blink, it means the system is currently communicating with ISP and has yet to obtain an IP address.**

✓ **When the WAN is connected, the WAN indicator will lights up in green and if data is being transmitted over the Internet, the indicator blinks in green and orange.**

Rear Panel

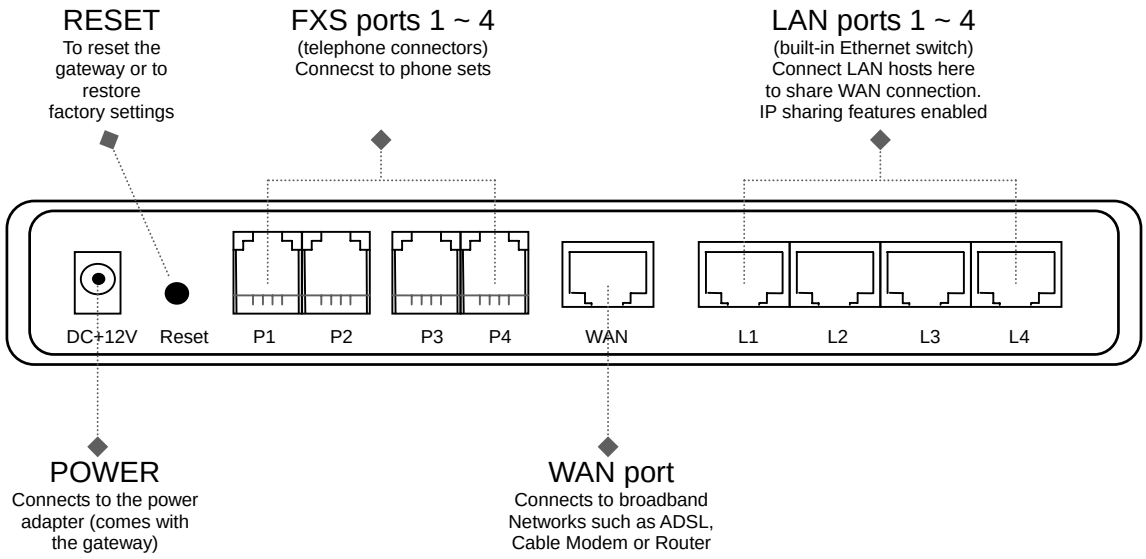
4S **:** P1-P4 are telephone ports (FXS) to be connected to a telephone set or PBX co ports (PSTN ports).
4O **:** P1-P4 are line ports (FXO) to be connected to a PSTN (telephone lines from a PSTN.)
2S2O **:** P1-P2 are telephone ports (FXS), and P3-P4 are line ports (FXO). Before the power is connected or in the occasion of a power failure, P1 will be relayed to P3, and P2 to P4 to reach PSTN.
3S1O **:** P1-P3 are telephone ports (FXS), and P4 is a line port (FXO). Before the power is connected or in the occasion of a power failure, P1 will be relayed to P4 so that emergency calls can be made.

NOTE: Do not connect FXS ports to each other. Also, do not connect any FXS port directly to a PSTN line or internal PBX. If any of these are done, your 4S may be damaged.

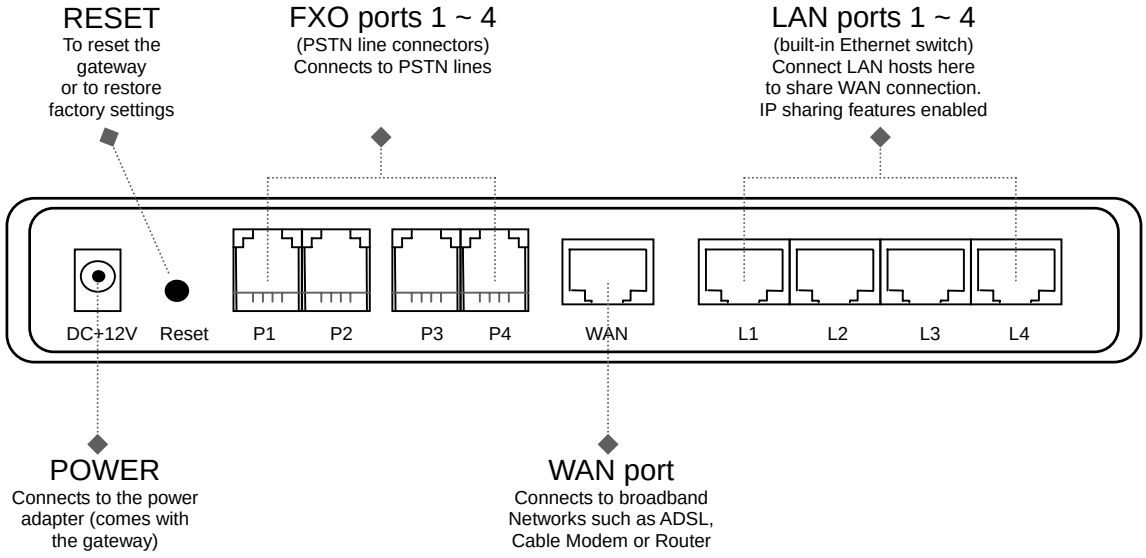
Please use the power adapter comes with VoIP gateway. If using adapters other than the one comes with VoIP gateway, it may cause problems and will affect the warranty of the product.

Restore to factory default: (IP address, User's. Name and Password)
(1) Pull off the power plug.
(2) Press reset (do not let go of the reset button).
(3) Plug the plug back into the socket (do not let go of the reset button).
(4) Let go of the reset button after 6 seconds. Factory settings will be restored.

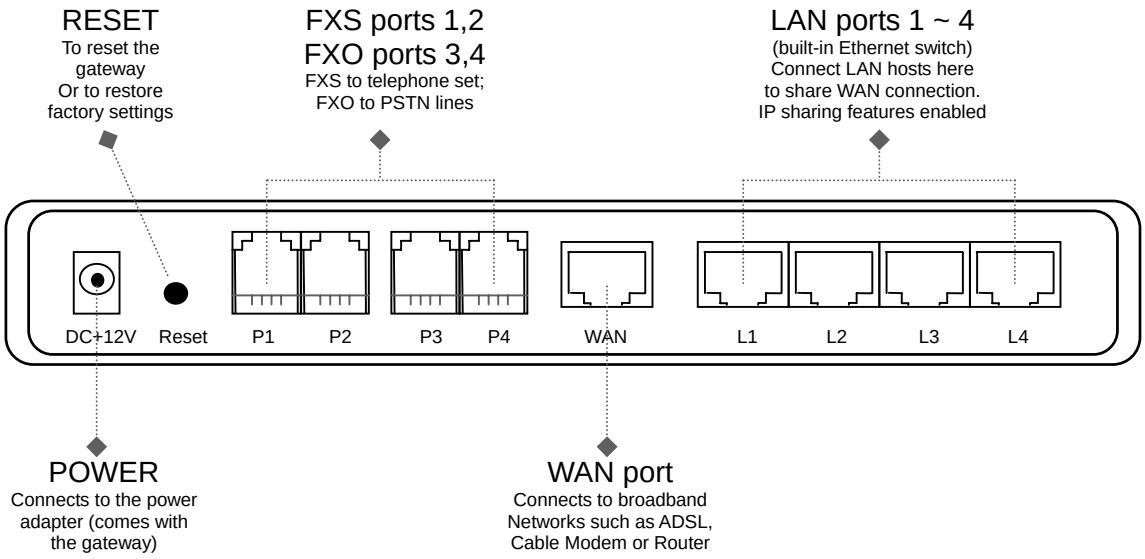
4S Model (4 FXS ports)



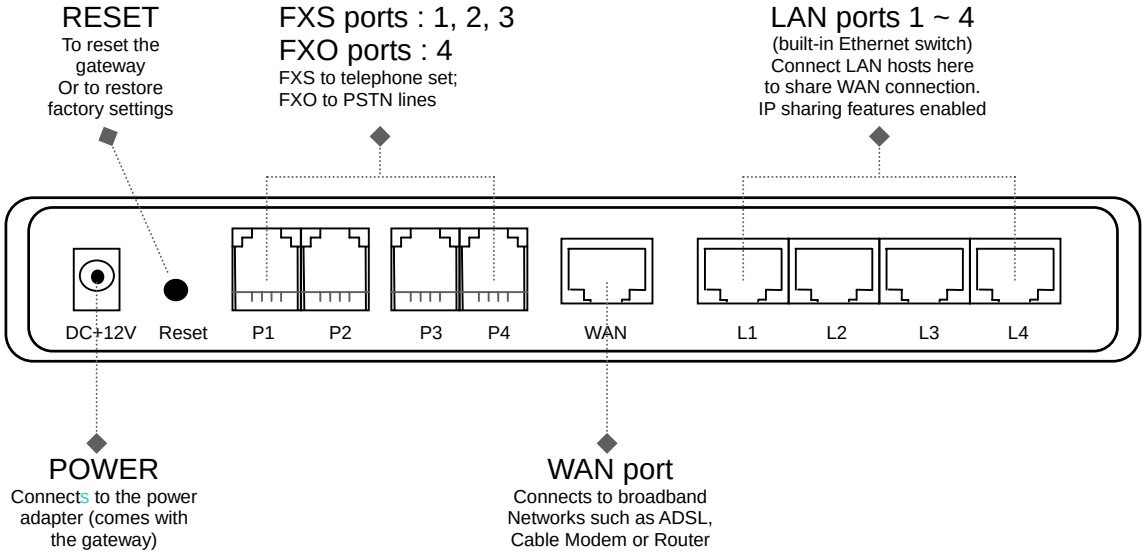
4O Model (4 FXO ports)



2S2O Model (2 FXS and 2 FXO ports)

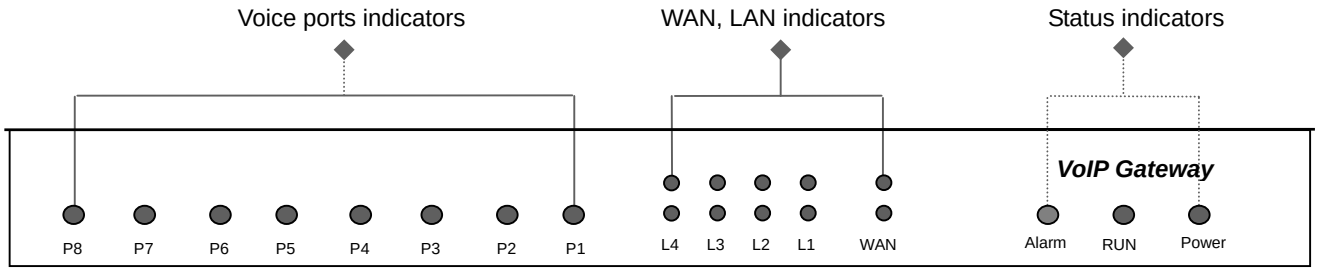


3S1O Model (3 FXS and 1 FXO ports)



8 ports gateways model: 8S / 8O / 6S2O / 4S4O

Front Panel



- Power Indicator: Green light indicates a normal power supply.
- Run Indicator: Blinking green light indicates normal operation.
- Alarm Indicator: When the system starts up, the red light will blink. It also indicates the gateway's abnormal operation.
- P1 – P8 stands for Port 1 – Port 8 (Line 1 – Line 8)
- WAN stands for the WAN Port Indicator.
- L1 – L4 stands for the LAN Port Indicator.

✓ **When starting up the system, the Alarm, Run, and Power indicators will light up. After about 15 seconds, the Alarm indicator will go off, the Run indicator will blink in green, and the Power indicator will stay green under operational conditions. If the Alarm indicator continues to blink, it means the system is currently communicating with ISP and has yet to obtain an IP address.**

✓ **When the ethereal cable is connected, the Link indicator will light up and if data is being transmitted over the Internet, the act indicator will blink.**

Rear Panel

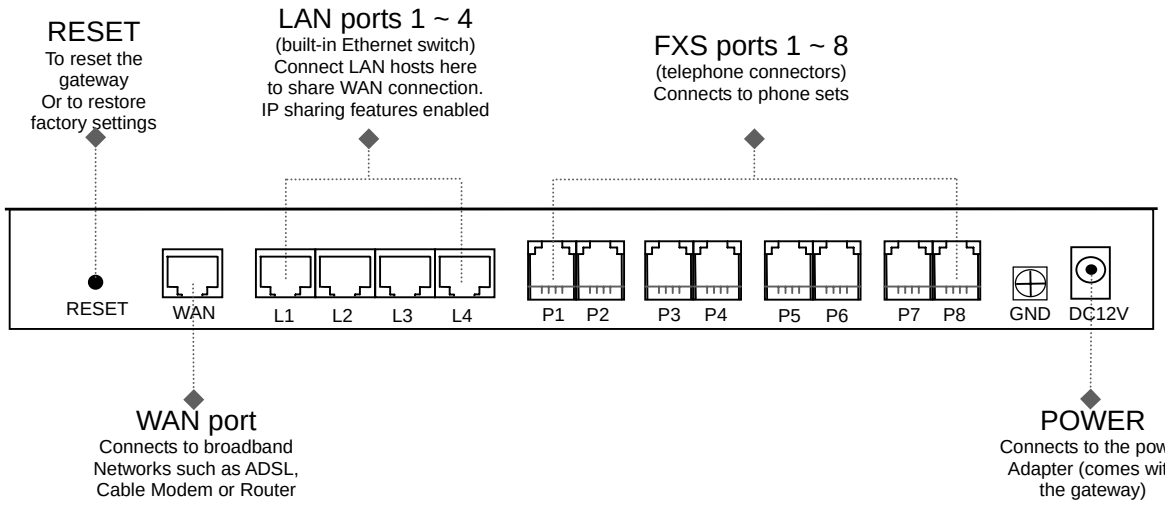
8S **:** P1-P8 are telephone ports (FXS) to be connected to a telephone set or PBX CO ports (PSTN ports).
8O **:** P1-P8 are line ports (FXO) to be connected to a PSTN (telephone lines from a PSTN.)
6S2O **:** P1-P6 are telephone ports (FXS), and P7-P8 are line ports (FXO). Before the power is connected or in the occasion of a power failure, P1 will be relayed to P7, and P2 relayed to P8 to reach PSTN.
4S4O **:** P1-P4 are telephone ports (FXS), and P5-P8 are line ports (FXO). Before the power is connected or in the occasion of a power failure, P1 will be relayed to P5, P2 relayed to P6, P3 relayed to P7, and P4 relayed to P8 to reach PSTN.

NOTE: Do not connect FXS ports to each other. Also, do not connect any FXS port directly to a PSTN line or internal PBX. If any of these are done, your 4S may be damaged.

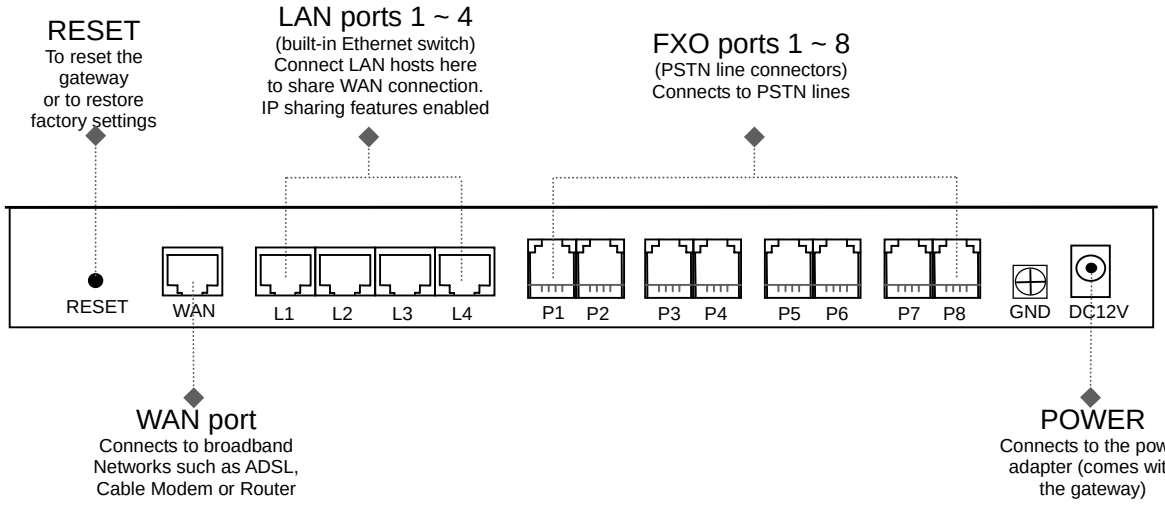
Please use the power adapter comes with VoIP gateway. If using adapters other than the one comes with VoIP gateway, it may cause problems and will affect the warranty of the product.

Restore to factory default: (IP address, User's. Name and Password)
(1) Pull off the power plug.
(2) Press reset (do not let go of the reset button).
(3) Plug the plug back into the socket (do not let go of the reset button).
(4) Let go of the reset button after 6 seconds. Factory settings will be restored.

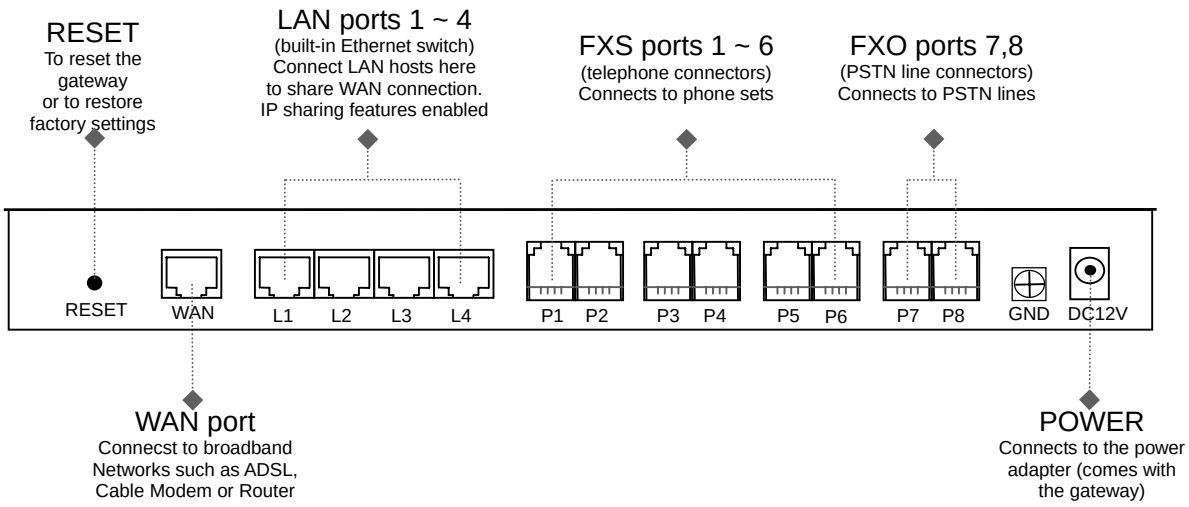
8S Model (8 FXS ports)


RESET
To reset the gateway Or to restore factory settings
LAN ports 1 ~ 4
(built-in Ethernet switch) Connect LAN hosts here to share WAN connection. IP sharing features enabled
FXS ports 1 ~ 8
(telephone connectors) Connects to phone sets
RESET
WAN
L1 L2 L3 L4
P1 P2 P3 P4 P5 P6 P7 P8
GND
DC12V
WAN port
Connects to broadband Networks such as ADSL, Cable Modem or Router
POWER
Connects to the power Adapter (comes with the gateway)

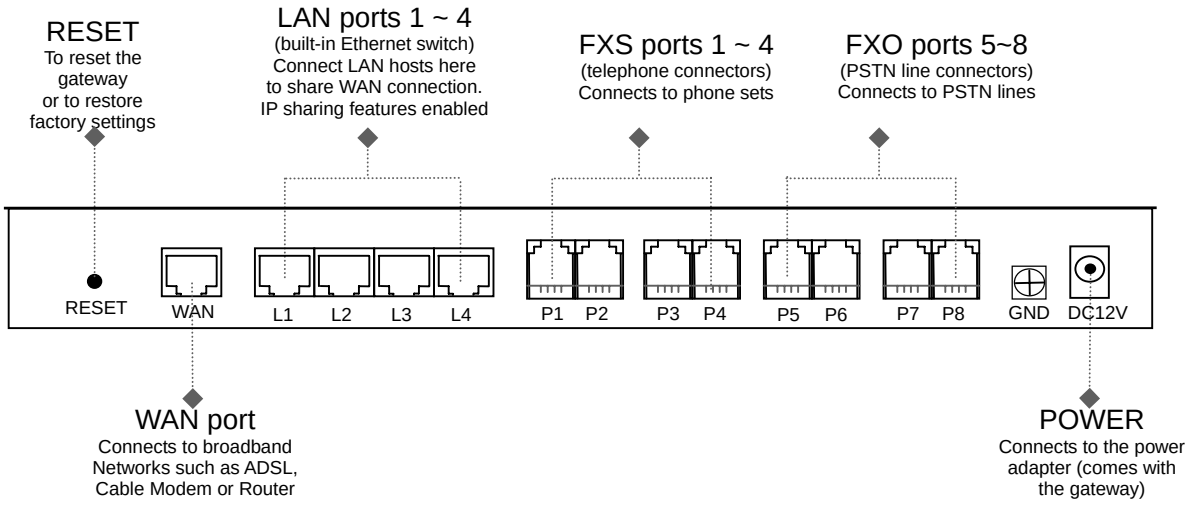
8O Model (8 FXO ports)


RESET
To reset the gateway or to restore factory settings
LAN ports 1 ~ 4
(built-in Ethernet switch) Connect LAN hosts here to share WAN connection. IP sharing features enabled
FXO ports 1 ~ 8
(PSTN line connectors) Connects to PSTN lines
RESET
WAN
L1 L2 L3 L4
P1 P2 P3 P4 P5 P6 P7 P8
GND
DC12V
WAN port
Connects to broadband Networks such as ADSL, Cable Modem or Router
POWER
Connects to the power adapter (comes with the gateway)

6S2O Model (6 FXS and 2 FXO ports)


RESET
To reset the gateway or to restore factory settings
LAN ports 1 ~ 4
(built-in Ethernet switch)
Connect LAN hosts here to share WAN connection.
IP sharing features enabled
FXS ports 1 ~ 6
(telephone connectors)
Connects to phone sets
FXO ports 7,8
(PSTN line connectors)
Connects to PSTN lines
RESET
WAN
L1 L2 L3 L4
P1 P2 P3 P4 P5 P6 P7 P8
GND
DC12V
WAN port
Connecst to broadband Networks such as ADSL, Cable Modem or Router
POWER
Connects to the power adapter (comes with the gateway)

4S4O Model (4 FXS and 4 FXO ports)


RESET
To reset the gateway or to restore factory settings
LAN ports 1 ~ 4
(built-in Ethernet switch)
Connect LAN hosts here to share WAN connection.
IP sharing features enabled
FXS ports 1 ~ 4
(telephone connectors)
Connects to phone sets
FXO ports 5~8
(PSTN line connectors)
Connects to PSTN lines
RESET
WAN
L1 L2 L3 L4
P1 P2 P3 P4 P5 P6 P7 P8
GND
DC12V
WAN port
Connects to broadband Networks such as ADSL, Cable Modem or Router
POWER
Connects to the power adapter (comes with the gateway)

2. Installation and Applications

Network Interface

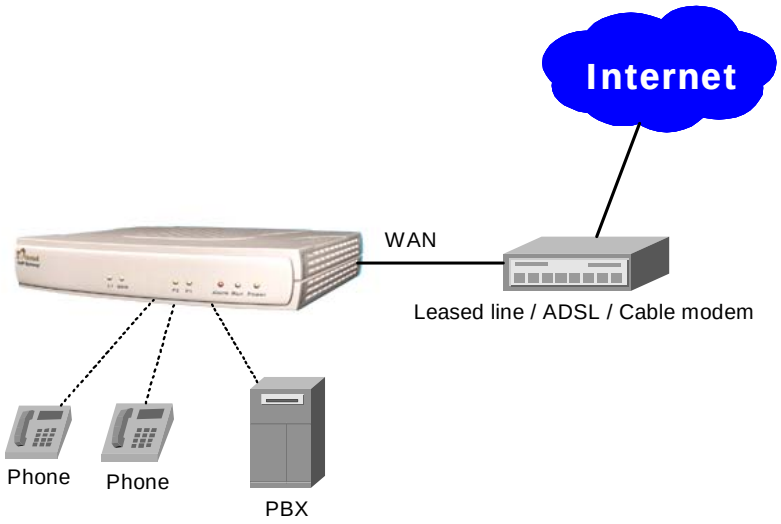
The network interface is divided into 3 basic modes as described below:

- Gateway can be assigned with a Public IP Address
- Gateway can be built under the existing NAT
- Gateway can be assigned with a Public IP address and serves as an IP sharing router.

Gateway Assigned with a Public IP Address

The gateway will have a Public IP address for Internet connection regardless of whether it is a static IP address, DHCP (using a Cable Modem), or PPPoE (Dialup / ADSL).

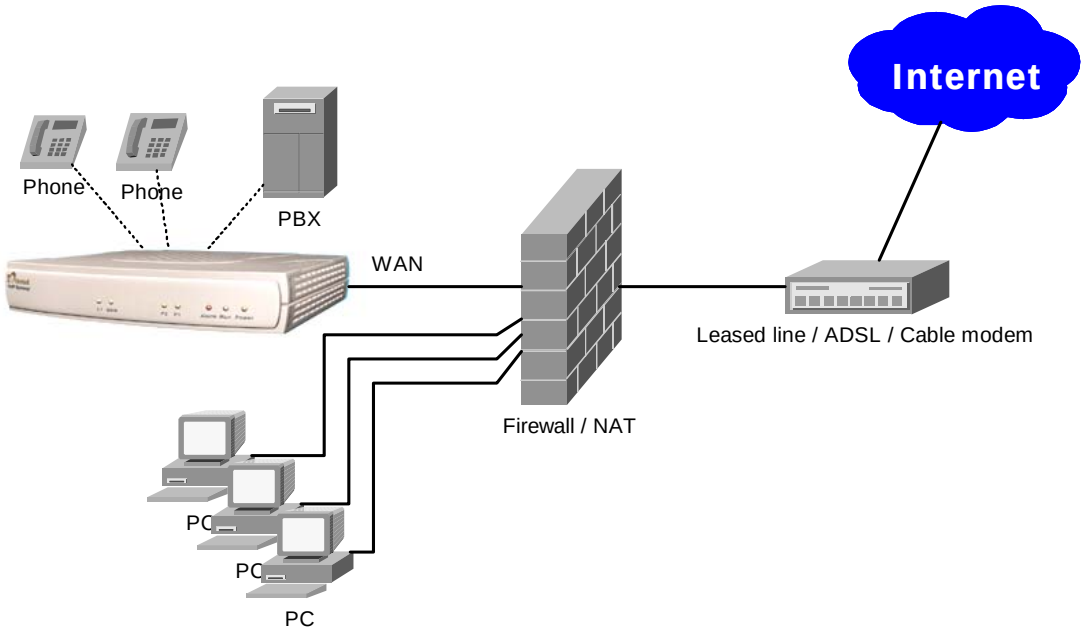
Gateway IP Settings	Need to be set up as static IP, DHCP, or PPPoE	
NAT/STUN Settings	Unnecessary (Disabled)	
DDNS Settings	Unnecessary (Disabled)	



Gateway in a NAT network

The gateway uses a virtual IP address and the IP sharing function of other systems to connect to the Internet.

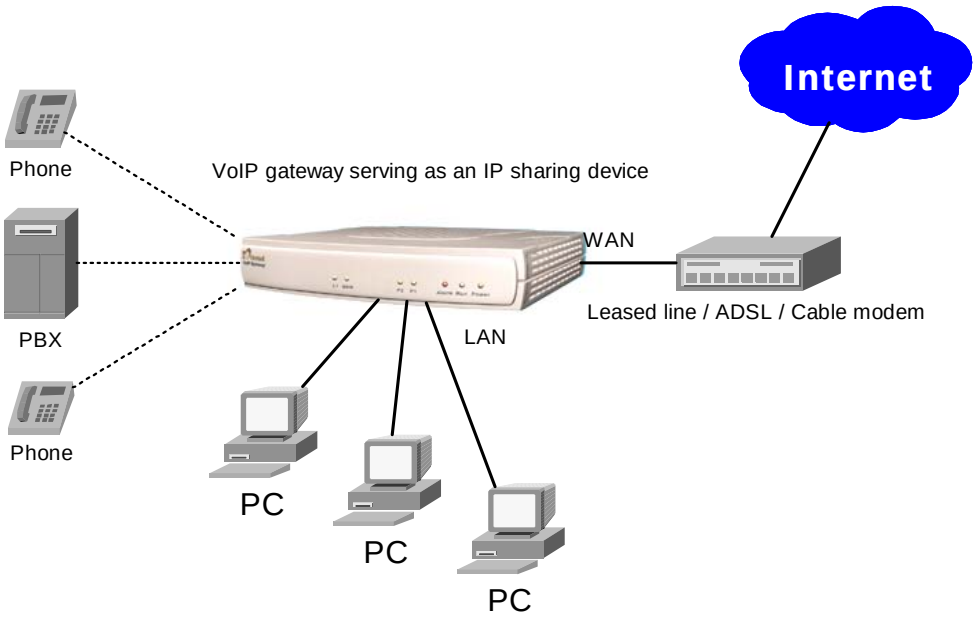
LAN IP address of IP sharing	Please avoid IP address 192.168.0.1-192.168.8.254 (You may need to change the settings of IP sharing or change SIP series Gateway LAN Port IP address)		
Gateway IP Settings	Set as static IP address, and assign the LAN IP address of the IP sharing to the Default Gateway.		
NAT /STUN Settings Please refer to P. 29 for the Use NAT.	Enable	If the WAN of the IP sharing device has static IP address, then the NAT IP address is set as the Public IP address of the IP sharing.	
		If the WAN of the IP sharing device uses a dynamic IP address, then it has to comply with the DDNS settings. When suing NAT, you must enter the URL (Uniform Resource Locator) that is registered to the DDNS server.	
DDNS Settings Please refer to P. 29 for the DDNS Settings	The WAN of the IP sharing device has a static IP address.		Disabled
	The WAN of the IP sharing device has a dynamic IP address.		Enabled: enter the registered URL (Uniform Resource Locator) into the network settings -> under NAT



Gateway assigned with a Public IP Address and serving as an IP sharing device

The gateway will have a Public IP address regardless of whether it is a static IP application, DHCP (using a Cable Modem), or PPPoE (To connect to your ADSL account), which can then use the functions of built-in IP sharing function to allow other PCs to be on-line at the same time.

Gateway IP Settings	Need to be set up as static IP, DHCP, or PPPoE	
NAT/STUN Settings	Unnecessary (Disabled)	
DDNS Settings	Unnecessary (Disabled)	
For settings at PC end, please refer to section "IP sharing functions"	PC uses a static IP address ranging from: 192.168.8.1-192.168.8.253 Subnet Mask：255.255.255.0 Default Gateway：192.168.8.254	

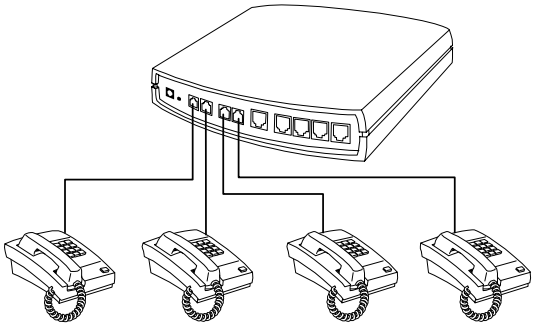


Telephone Interface Description (4 ports model used in the example)

Example for 4S gateway:

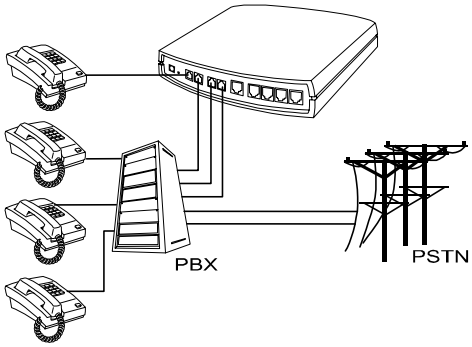
4S gateway connecting directly to phone sets

After connecting telephone sets to P1-P4, users can make direct calls,(P1-P4 are FXS interfaces). Each set acts as an independent extension line.



Integrating the 4S with PBX

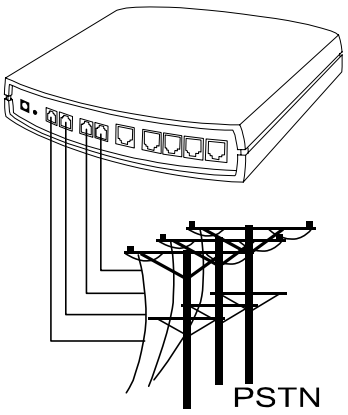
P1-P4 is FXS interfaces, and some of them can be connected to telephone sets for direct calls. Others can be connected to the PBX so other extension lines can make VoIP calls.



Example for 4O gateway:

4O model connecting directly to the Telephone Line of a PSTN

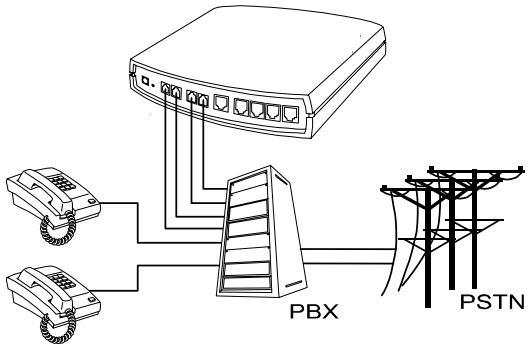
P1-P4 is FXO interfaces and can all be connected to a PSTN to serve as a bridge between the PSTN and other VoIP telephones. The system also allows a call to be made from a traditional telephone line to connect with a user behind the Gateway.



Integrating the 4O with PBX

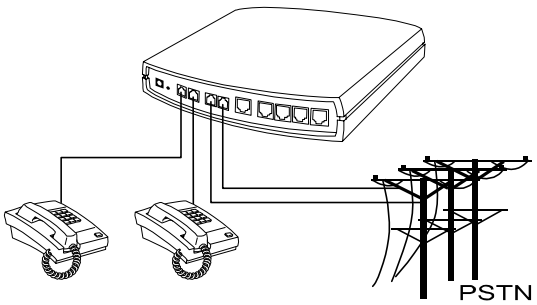
P1-P4 is FXO interfaces and can be connected with PBX extension lines (exclusively for analog interface, not applicable for digital type).

•



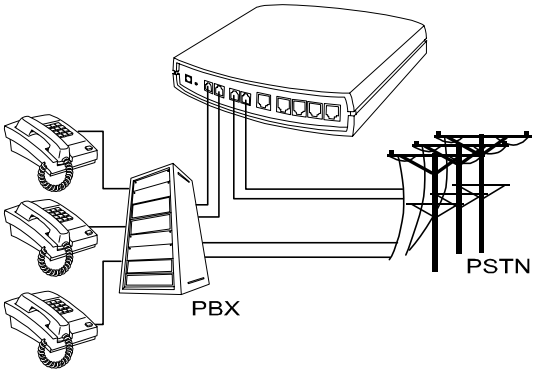
Example for 2S2O gateway:

P1-P2 is FXS interfaces and can be directly connected to a telephone set for direct calls. P3-P4 is FXO interfaces and can be connected to a PSTN to serve as a bridge between the PSTN and other VoIP telephones. The system also allows a call to be made from a traditional telephone line to connect with a Gateway user.



Integrating the 2S2O with PBX

P1-P2 is FXS interfaces and can be connected to a PBX CO; P3-P4 is FXO interfaces and can be connected to a PSTN to act as a bridge between the PSTN and other VoIP telephones. The system also allows a call to be made from a traditional telephone line to connect with a Gateway user.



3. Setting the Gateway through IVR

VoIP transmits voice data (packet) via the Internet to achieve telecommunications. This means that the telecommunication quality is closely related to the whole network environment. If any one of the telecommunicating parties has insufficient bandwidth or frequent packet loss, the telecommunication quality will be poor. Therefore, an excellent telecommunication can only be created when Gateway is connected to the Internet and when network environment is stable.

Preparation

- Install the Gateway according to instructions. Connect the power supply, telephone set, telephone cable, and network cable properly as described in Chapter 2.
- If a static IP is used, confirm the desired IP settings of the WAN Port (IP address, Subnet Mask, and Default gateway). Please contact your local Internet Service Provider (ISP) if you have any questions.
- If using dialup ADSL (PPPoE) for network connection, confirm the dialup account number and password.
- If users wish to build Gateway under the NAT, Gateway WAN Port IP address and LAN Port should not use the same range. This is to avoid phone failures.

Basic Settings of a Gateway

- IP Settings— Connecting Gateway to the Internet.
- Telephone Number of Gateway Phone Number (

Gateway representative number).
- Using **Phone Book** or **Phone books manager**

Please refer to Section 6 for Basic Dialing Method

Gateway provides two setting modes: 1. Telephone IVR Setting Mode, and 2. Browser Setting Mode. The IVR provides basic query and setting functions, while the browser provides a full setting function.

IVR (Interactive Voice Response)

Gateway provides convenient IVR functions. Users only need to pick up a handset and enter the function code for the query and setting without using a PC.

NOTE: After finishing the settings, make sure the new settings are saved. This is so that the new settings will take effect after the system is restarted.

Instructions

- FXS Port: Connected to telephones. To enter IVR mode, enter " * * password #" after hearing the dial tone. When you hear a second dial tone, the system is in IVR mode, enter the function code. (Please refer to the Advanced Settings on page 35 for these codes)

Example: The factory default code is blank. Enter **"**#"**. You are now in IVR setting mode, enter the desired code. E.g.: if the code is 1234, then enter **"**1234#"**.

- FXO Port: to use IVR functions, dial the phone number of FXO Port using an external line. You will hear the instruction "enter value", and then enter a PIN number. The factory default code is blank. Enter "**#" as above. You are now in IVR setting mode.
- Once the first setting or query has been completed, you will hear a dial tone. Then use the same procedure to make a second query or setting. To exit IVR mode, simply hang up the phone.

Example: enter **"**#"** (You are now in IVR mode)→ enter **101** (to query IP address) → the system responds with an IP address →you can continue with more settings or queries: enter **111** (to set IP address) →enter **192*168*1*2** (IP number).

Save Settings

After entering IVR mode, dial 509 (Save Settings). Wait for about 3 seconds and after hearing a confirmation tone "1", hang up the phone. Please reboot the Gateway to enable the new settings.

To inquire about current Gateway's WAN Port IP address

After entering IVR mode, dial 101. The system will repeat the current WAN Port IP address.
If the system does not repeat the IP address, it indicates that the Gateway is not currently connected to the Internet. Please check if the cable connection, account number, and password are correct.

Software Upgrades

IVR provides online upgrades. Once in IVR mode, enter "209" and you will hear "Enter Value". Enter your IP address followed by "#" (ie.：**61*30*25*89#**). Then you will hear a second "Enter Value". Enter the Listen Port Number followed by "#"(ie.: **6001#**). For information about upgrading, please ask your agent.

IVR Functions Table:

Function Code	Description	Example
111/101	WAN Port IP address Set/Query	Use in conjunction with function code **114**, select 1 for a Static IP function.
112/102	WAN Port Subnet Mask Set/Query	
113/103	WAN Port Default Gateway Set/Query	
114/104	Current Network IP Access Set/Query (1: Static IP, 2.DHCP, 3.PPPoE)	
115/105	DNS IP address Set/Query	
116/106	Phone books manager IP address Set/Query	Must use 116/106, 117/107 in conjunction with each other.
117/107	Set/Query whether or not to use Public Telephone Book (0: Disable 1:Enable)	
199/099	Set/Query whether or not this Gateway acts as the phone books manager (0: Disable 1: Enable)	
066	Querying the connection to Phone books manager	
118	Restart	
121	Setting PPPoE Account	Use in conjunction with function code **114,** select 3 for a PPPoE function
122	Setting PPPoE Password	
123	Setting NAT IP address	Must use 123 and 124 in conjunction with each other.
124	Uses NAT (0: Disable 1: Enable)	
151/141	Register to Proxy Server Set/Query (0: Disable 1: Enable)	
152/142	Proxy Server IP address Set/Query	
153/143	Proxy Server Port Set/Query	
125	Set Proxy Server account	
126	Set Proxy Server password	
154/144	Uses STUN Set/Query (0: Disable 1: Enable)	
155/145	STUN IP address Set/Query	
156/146	STUN Port Set/Query	
311/301		
312/302		
131/132		
133	Saving greeting message	
211/201	Set/Query International Prefix code	Prefix dialed before making an international call e.g. 002 and 005.
212/202	Set/Query Country Code	Setting country code, e.g. 886
213/203	Set/Query Area Prefix Code (Long-Distance Prefix Code)	Prefix dialed before making a long-distance call e.g. 0.
214/204	Set/Query Area Code	eg. “2” for Node B area.

Function Code	Description	Example
215/205	Set/Query Gateway Telephone Number (Representative Number)	
216/206	Set/Query the extension number of Line 1.	
217/207		
109	Restoring factory default IP address configuration	A static IP address for WAN Port IP：192.168.1.2 Mask：255.255.255.0 Gateway：192.168.1.254
409	Restoring factory default settings	
509	Save settings	
900	Setting IVR and the language used on the Web GUI (1: English, 2: Traditional Chinese, 3: Simplified Chinese)	
209	Soft Upgrade	

IP Configuration Settings—Setting IP Configuration of WAN Port

Static IP Settings

NOTE: Complete static IP settings should include a static IP (Option 1 under114), IP address (111), Subnet Mask (112), and Default Gateway (113). Please contact your local Internet Service Provider (ISP) if you have any questions.

Function	Command
Select a Static IP	• After entering IVR mode, dial 114. • After hearing "Enter value", dial 1 (to select static IP)
IP address Settings	• After entering IVR mode, dial 111. After hearing "Enter value", enter your IP address, followed by "#". Example: If the IP address is 192.168.1.200, dial 192*168*1*200#.
Subnet Mask Settings	• After entering IVR mode, dial 112. After hearing "Enter value", enter your subnet mask, followed by "#". Example: If the mask value is 255.255.255.0, dial 255*255*255*0#.
Default Gateway Settings	• After entering IVR mode, dial 113. After hearing "Enter value", enter your default gateway's IP address, followed by "#". Example: If the Default Gateway is 192.168.1.254, dial 192*168*1*254#.
Save Settings and Restart	• To save settings, dial **509** (Save Settings). The system will save the settings. Please restart the system. Wait for about 40 seconds for the system to restart, and then enter **101** to check if the IP address is retained. If the IP address is not repeated, it indicates that Gateway has not been properly connected, please check if the cable connection, account, or password are correct.

Dynamic IP (DHCP) Settings

After entering IVR mode, dial 114.
After hearing "Enter value", dial 2 (to select DHCP).
Saving settings –press 509 (Save Settings). Please restart the system. After the system is restarted, press 101 to check if the IP address is retained.
If the system does not repeat the IP address, it indicates Gateway has not been properly connected to the Internet. Please check the cable connection.

ADSL PPPoE Settings

NOTE: Complete PPPoE settings should include: Select PPPoE (Option 3 of 114), PPPoE account (121) and PPPoE password (122).

Please contact your local Internet Service Provider (ISP) if you have any questions.

Select a PPPoE

- After entering IVR mode, dial 114.
- After hearing "Enter value", dial 3 (to select PPPoE).

PPPoE Account Settings

- After entering IVR mode, dial 121.
- After hearing "Enter value", enter the account number, followed by "#".

Example: If the account is "84943122 @ hinet.net", please enter 08040904030102027148495445607254 4560#.

Please note that it is necessary to enter two digits for each character/number; for example, enter "01" for "1" and "11" for "A".

PPPoE Password Setting

- After entering IVR mode, dial 122 after hearing "enter value" followed by "#".

Example: If the password is "3ttixike", please enter "03 60 60 49 64 49 51 45#".

Save Settings and Restart

To save settings, dial **509** (Save Settings). The system will save the settings. Please restart the system. Wait for about 40 seconds for the system to restart, and then enter **101** to check if the IP address is retained. If the IP address is not repeated, it indicates that Gateway has not been properly connected, please check if the cable connection, account, or password are correct.

Recorded Voice File

- The gateway allows users to record their incoming call greeting messages, when calling via FXO.
- After entering IVR mode, dial 132. After hearing "Enter value", record the incoming call greeting message. To end recording, simply hang up.
- After recording, to listen to the recorded message, press 131. Press 133 to save the message.

PPPoE Character Conversion Table

Number	Input Key	Upper Case Letter	Input Key	Lower Case Letter	Input Key	Symbol	Input Key
0	00	A	11	a	41	@	71
1	01	B	12	b	42	•	72
2	02	C	13	c	43	!	73
3	03	D	14	d	44	"	74
4	04	E	15	e	45	$	75
5	05	F	16	f	46	%	76
6	06	G	17	g	47	&	77
7	07	H	18	h	48	'	78
8	08	I	19	i	49	(	79
9	09	J	20	j	50	)	80
		K	21	k	51	+	81
		L	22	l	52	,	82
		M	23	m	53	-	83
		N	24	n	54	/	84
		O	25	o	55	:	85
		P	26	p	56	;	86
		Q	27	q	57	<	87
		R	28	r	58	=	88
		S	29	s	59	>	89
		T	30	t	60	?	90
		U	31	u	61	[	91
		V	32	v	62	\	92
		W	33	w	63	]	93
		X	34	x	64	^	94
		Y	35	y	65	_	95
		Z	36	z	66	{	96
						|	97
						}	98

4. Setting a Gateway with WEB Browser

The gateway allows users to make settings using a web browser. After opening a browser, enter Gateway's IP address as the website address in order to enter the Web configuration screen as shown in the following diagram. (IE Browser used for example: Enter http：//192.168.1.2.)
The factory default WAN IP address for Gateway is **192. 168. 1. 2**.You can also enter "101" from the handset to inquire about the current WAN Port IP address. The factory default LAN Port IP address is 192.168.8.254.

Instructions

- Open an Internet browser.
- Enter gateway's WAN Port IP address in the website address area (If the PC is connected to the LAN Port, enter the LAN Port IP address. The default is 192.168.8.254)
- The following registration screen will appear (The factory default settings for **Login ID and Password are set to blank**).
- After completing and confirming the settings, some of the settings will take effect immediately. But network related settings would take effect after the gateway is restarted. Please go to **System Operation** to save the settings before restarting the system.

To avoid several people simultaneously configuring the web and causing problems to users, please enter the correct Login ID and password. If a user logs into the system, other users from different IP addresses cannot login at the same time. Please remember to logout or restart the system if not using the web configuration function.

Network Settings

The network settings are used to set the gateway's communication ports, IP configurations, and Phone Books Manager IP etc.

Network Settings (WAN)			
Current WAN IP Address	192.168.1.3		
Listen Port UDP [1 - 65535]	5060	RTP Starting Port UDP [1 - 65500]	9000

DHCP ○		
Static IP ◉	IP address	192.168.1.2
	Subnet mask	255.255.255.0
	Default Gateway IP	192.168.1.254
PPPoE ○	PPPoE Account	
	PPPoE Password	
	Confirm Password	
PPTP ○	IP address	
	Subnet mask	
	PPTP Server	
	PPTP ID	
	PPTP Password	
	Confirm Password	
BigPond Cable ○	User Name	
	BigPond Cable Password	
	Confirm Password	
	Login Server	
Domain Name Server Assignment	○ Auto ◉ Manual	
Domain Name Server (Primary) IP	192.168.1.254	Domain Name Server (Secondary) IP

- Current WAN IP Address: The IP address of WAN port.
- Listen Port UDP: It is not necessary to change the protocol of the communication port used by VoIP gateway.

- RTP Starting Port UDP: The initial value of port number for transmitting voice data among Gateway(s). Each line requires 2 ports. It is not necessary to change these.

For example, If the starting port is 9000, then Line 1 is using 9000 and 9001, and Line 2 is using 9002 and 9003, and so forth.

IP Configuration (Setting WAN Port)

There are four methods of obtaining a WAN port IP address:
1. Static IP
2. DHCP, means a Dynamic IP (Cable Modem)
3. PPPoE (Dialup ADSL)
4. PPTP.

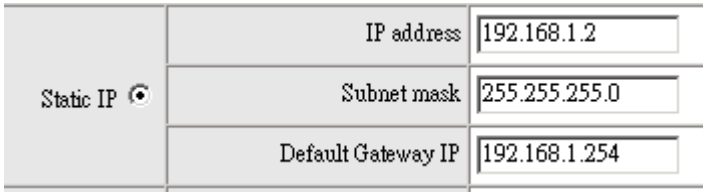
Using the DHCP and PPPoE for obtaining an IP address may vary. If not familiar with the network connection, please contact your local ISP.

Setting Dynamic IP (DHCP)

Click "DHCP" to obtain a Dynamic IP address, then click the "Accept" button at the bottom of the screen.
Save the settings: Click **System Operation** to select "Save Settings",
"Restart", and then click the "Accept" button. Wait for a while (about 40 seconds), and the system will obtain the related IP value from the DHCP Server.

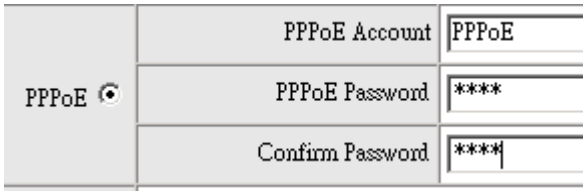
NOTE: After the system has obtained a new IP address, if using WAN Port to enter the Web Configuration Screen, a new IP address has to be used. The same applies to the following two settings.

Setting Static IP



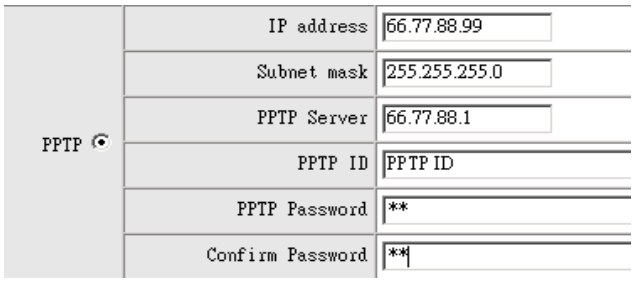
Select "Static IP" and enter the IP address, Subnet Mask and Default Gateway values. Then click the "Accept" button at the bottom of the screen.
Save the settings, and then restart the system. Wait for about 40 seconds for the system to restart.

ADSL PPPoE Settings



Select "PPPoE" Enter the Account Number, Password and Reenter Password to confirm. Then click the "Accept" button at the bottom.
Save the settings, and then restart the system. The system will take about 49 seconds to restart.

PPTP※

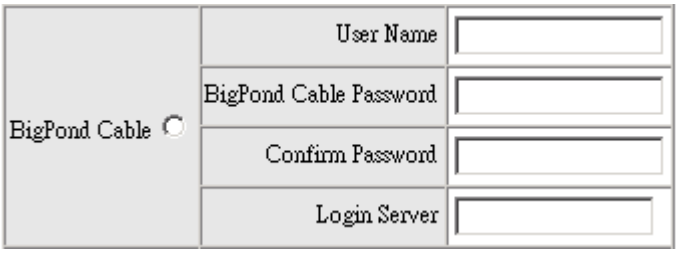


Select "PPTP" and enter the IP Address, Subnet mask, PPTP Server, PPTP ID and Password. Then click the "Accept" button at the bottom.

Save the settings, and then restart the system. The system will take about 40 seconds to restart.

If not familiar with the network connection, please contact your local ISP.Domain Name Server

BigPond (for Australia use only)



Click "BigPond Cable" Enter User Name and Password. Login Server is optional. Then click the "Accept" button at the bottom.

(DNS) Settings

Domain Name Server (DNS): While a gateway is accessing another gateway or computer with a hostname, it will look up the IP address from the DNS provided by ISP. The ISP whilst negotiating with PPPoE or DHCP usually assigns the DNS information. In the case that the DNS is not assigned automatically or WAN port is assigned with a static IP address, the DNS information must be assigned manually.

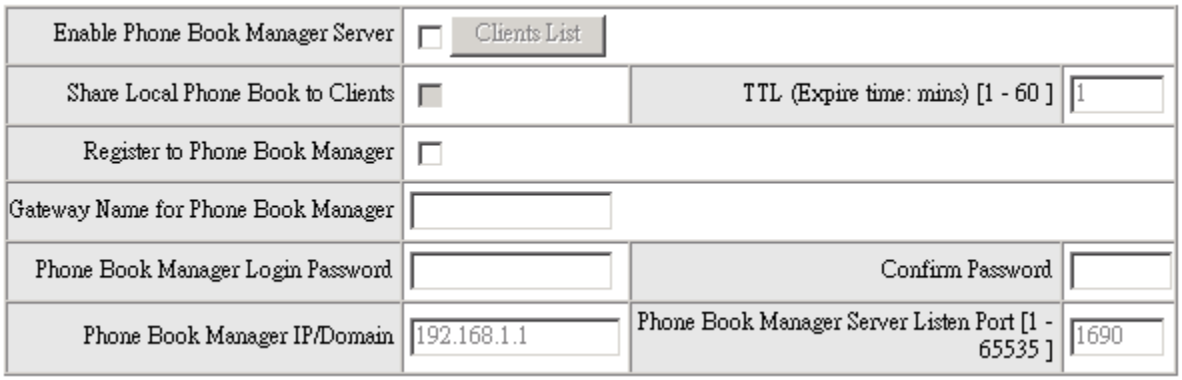
Auto : Gateway learns primary & secondary addresses from ISP's DHCP server or PPPoE server.

Manual : Enter the primary & secondary addresses manually. Please be sure the IP addresses are correct otherwise the gateway will not be able to access hosts with its hostname.

Clone MAC

Some Internet Service Providers (ISP) assigns the bandwidth via the MAC (Media Access Control) Address. You can click the " Clone" button to copy the MAC address of the Ethernet Card installed in the computer used to configure the device. It is only necessary to fill in the field if required by your ISP. The "Your MAC Address" will be blank as you log in through WAN port.

Using Phone Books Manager


Enable Phone Book Manager Server
Clients List
Share Local Phone Book to Clients
TTL (Expire time: mins) [1 - 60]
1
Register to Phone Book Manager
Gateway Name for Phone Book Manager
Phone Book Manager Login Password
Confirm Password
Phone Book Manager IP/Domain
192.168.1.1
Phone Book Manager Server Listen Port [1 - 65535]
1690

- Enable Phone Books Manager Server: It allows other Gateway users to register the IP address and telephone number in this Phone books manager. It is recommended that the unit appointed as the Phone Book Manager use static IP.
- Share Local Phone Book: While this option is enabled and the gateway is performing as a Phone Books Manager, this gateway will append its local Phone Book entries to the Manager for other clients to lookup.
- TTL (Time to Live): If a Gateway system that is controlled by the Phone Books Manager does not report back within the deadline set by TTL, the system will be excluded from the user's list. Each Gateway should report to the Phone Books Manager once every 30 seconds.
- Register to Phone Books Manager: To register to the Phone Books Manager.
- Gateway Name for Phone Book Manager: The alias registered with the Phone Books Manager.
- Phone Books Manager Login Password: Enter the registered password. If this system is serving as the Phone Books Manager, the set password is also the password used for registering other Gateway systems.
- Phone Books Manager IP/Domain: Enter the IP address for the Phone Books Manager. It supports URL (Uniform Resource Locator).
- Phone Books Manager Port: The protocol communication port for transmitting signals between the Phone Books Manager and other Gateway systems. Please confirm whether the setting is the same as that of the **Phone Books Manager.**

LAN interface mode
◉ Router ○ Bridge

Network Settings (LAN)			
LAN IP / LAN default Gateway	192.168.8.254	Subnet mask	255.255.255.0
DHCP Server			
Enable DHCP Server	☐		
IP Pool Starting Address	192.168.8.1	IP Pool Ending Address	192.168.8.250
Lease Time [1 - 9999 hours]	1		
Domain Name Server Assignment	◉ Auto ○ Manual		
Domain Name Server (Primary) IP		Domain Name Server (Secondary) IP	

Port of Web Access from WAN [1 - 65535]	80

LAN interface mode※

- Router: The system serves as a router.
- Bridge: The system serves as a bridge between WAN port and LAN port.

Network Settings (LAN)

- Network Settings (LAN): Gateway LAN Port IP address and the subnet mask value. Please note that Gateway is built under NAT: Gateway LAN Port IP address cannot be in the same section as the NAT LAN Port IP address, or else it is unable to make or receive calls. For example, if the NAT LAN Port IP address is 192.168.8.1, then Gateway LAN Port cannot be ranged between 192.168.8.1 ~ 192.168.8.254. If so, please change the LAN port IP address, (e.g. setting the IP address to 192.168.99.254.)

DHCP Server			
Enable DHCP Server	☑		
IP Pool Starting Address	192.168.8.1	IP Pool ending Address	192.168.8.250
Lease Time[1 - 9999hours]	1		

- Enable DHCP Server: Enable or Disable DHCP server service of gateway.
- IP Pool Starting Address: The first IP address to be assigned to DHCP clients.
- IP Pool ending Address: The last IP address to be assigned to DHCP clients.
- Lease Time: The valid period of an assigned IP address.

Domain Name Server Assignment	◉ Auto ○ Manual		
Domain Name Server (Primary) IP		Domain Name Server (Secondary) IP	

- Domain Name Server Assignment: The DNS information to be assigned to DHCP clients.
  Auto : Assigns the same DNS information of WAN port to the DHCP clients.
  Manual : Manually assigns the DNS information for DHCP clients.

Port of Web Access from WAN[0 - 65535]	80

- Port of Web Access from WAN: Http port for WAN. To make this setting, the LAN Port must be used. Settings cannot be made using the WAN Port. Always use port 80 when connecting to LAN port.

QoS Settings

WAN QoS

WAN QoS		
☐ QoS	Downstream Bandwidth	Full
	Upstream Bandwidth	Full
ToS / DiffServ Settings		
ToS IP Precedence ◉	Signaling Precedence	3 (Flash)
	Voice Data Precedence	5 (CRITIC / ECP)
DiffServ (DSCP) ○	Signaling Value	26 (Assured Forwarding Class 3 - Low Drop Precedence, AF31)
	Voice Data Value	46 (Expedited Forwarding, EF)

- QoS (Quality of Service): Sets an external bandwidth to ensure sound quality during transmission (When this function is enabled, the voice packet has the highest priority to ensure telecommunication quality while less bandwidth is assigned for data transmission). Some models of the VoIP gateway without this function can adjust the bandwidth automatically.
- ToS/DiffServ (Type of Service/DSCP): The voice packet has the highest priority to ensure telecommunication quality; the larger the value you set, the higher priority you will get.

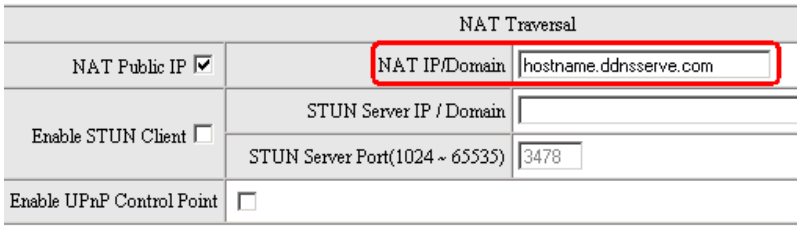
LAN QoS※

LAN QoS				
Enable LAN QoS	☐			
Port #	Priority	Flow Control	Incoming Rate Limit	Outgoing Rate Limit
LAN Port 1	Low	☐	Full	Full
LAN Port 2	Low	☐	Full	Full
LAN Port 3	Low	☐	Full	Full
LAN Port 4	Low	☐	Full	Full

- Priority: Set the priority of physic LAN port.
- Flow Control: Enable or Disable Flow control.
- Incoming Rate Limit: Set the incoming rate limit of a specific LAN port.
- Outgoing Rate Limit: Set the outgoing rate limit of a specific LAN port.

NAT/DDNS

NAT Traversal



If a Gateway is set up under an IP sharing setting, you can select either the NAT or STUN protocol.

- NAT Public IP: The IP address used by the gateway should be a virtual address. Further more, users must set the Virtual Server Mapping in the NAT Server (A virtual server is defined as a Service Port, and all requests to this port will be redirected to this specified the server IP address).

  The default port is listed below:
  Listen Port (UDP): 5060
  RTP Starting Port (UDP): 9000~9015 (Listen Port used for telephone communication).
  Http Port (TCP): 80

- NAT IP/Domain: Enter the NAT Server IP address (Real External IP address of NAT Server) then fill in the URL (Uniform Resource Locator). If using DDNS, please refer to the next setting item
- Enable STUN Client: Using STUN protocol prevents problems with setting the IP sharing function, but some NAT do not support this protocol.
- STUN Server IP/Domain and Port: Enter the STUN server IP address and Listen Port number. You can set 2 STUN server spread by semicolon.
- Enable UPnP Control Point: To enable the VoIP gateway's IP traffic to pass through a NAT server. This function only works when the NAT server supports UPnP and has it enabled.

DDNS

These settings are only necessary when the gateway is set up under a NAT that uses a dynamic IP address and do not support DDNS.
Choose a DDNS Server: The current system allows users to choose either DynDNS、TZO、3322.org、PeanutHull or a private server. Please apply for a user account before choosing a service provider.

- Server address: Sets up the IP address or URL (Uniform Resource Locator) of the DDNS Server.
- Hostname: The URL of the system (or NAT) – apply from a domain name registration providers.
- Login ID and Password: The ID and password are used to log into the DDNS server.
- Behind NAT: Select only when the system is set up under NAT.

NOTE: If the Gateway is set up under NAT, then enter the hostname into the NAT IP/Domain.

Telephony Settings

Prefix Number Rules

Prefix Number Rules	
Trunk Dial Out Verify	01;00
Trunk Dial Out Replace	190601;190200
Trunk Dail Out Deny	020

- Trunk Dial Out Verify/ Trunk Dial Out Replace: The system will transfer the number for all transit out call through FXO port. For example: If you transit out with 01907123456, the system will trans to 190601 907123456. If you transit out with 008621123456 the system will replace it with 190200 8621123456.
- Trunk Dial Out Deny: The system will deny the call with the leading number filled in this column.

FXS Caller ID Generation	◉ Disable	○ DTMF	○ FSK
FXO Caller ID Detection	☑	Detection Level	0
FSK Caller ID Type	◉ Bellcore	○ ETSI	
Anonymous Caller ID (CLIR)	☐		
Anonymous Transit in W/O Caller ID	☐		
Trunk Incoming Prompt Voice	◉ Default Greeting	○ Recorded voice file	○ Dial Tone

- FXS Caller ID Generation: Select this option to enable the caller ID display function. When enabled, the caller's phone number will be displayed on your phone set when the call comes through. FSK is preferred in some countries.
- FXO Caller ID Detection: To detect the Caller ID delivered from PSTN to the FXO port. While enabled, the Caller ID detected on the FXO port will be send to the SIP Proxy Server on dialing out calls.
- Detection Level: If FXO can't detect Caller ID, try to adjust it until it can.

  You have to enable **"**Wait for Caller ID before FXO / Trunk pick up" to ensure Caller ID is detected correctly**.**
- FSK Caller ID Type: Select FSK type. In most cases, Bellcore is preferred in North America and ETSI in Europe.
- Anonymous Caller ID (CLIR): When enabled, the caller's phone set will not display your number.

  If you register the VoIP gateway to a Proxy, you may be unable to make a call. This is due to the fact that the VoIP gateway doesn't send the number for authorization.
- Anonymous Transit in W/O Caller ID: FXO won't detect caller ID, and the gateway will dial out with an anonymous caller identficiation. If the call needs caller ID to be identified for Proxy, Proxy will reject this call without caller id.

- Trunk Incoming Prompt Voice: Select the greeting (must use the IVR 132 function to record a voice file) when FXO receives an inbound call.

	Enable	Type	Hot Line	Hot Line No.	Warm Line (Hot Line Delay)	Dial-Out Prefix	Group Hunting	Grounding Compensation	Enable FAX
Line 1	☑	FXS	☐				☑		☐
Line 2	☑	FXS	☐				☑		☐
Line 3	☑	FXO	☐		0 [0 - 60 s]			☐	☐
Line 4	☑	FXO	☐	701	0 [0 - 60 s]			☐	☐

- Enable: To enable a line; if some lines are not used, disable them (Pause Function) to avoid unnecessary waiting when an incoming call is diverting to this line.

Hotline Functions

- FXS port: When the user picks up the phone, the gateway automatically dials your assigned hotline number. When in hotline mode, other lines cannot be used.
- FXO port: When receiving a call from an outside line, the gateway will divert the call to the assigned hotline number.
- Hot Line No.: Enter the hot line number for an automatic dialing function.
- Warm Line: When the Warm Line function is in use, user can dial a number. Otherwise the system will divert incoming calls from an outside line to the Hot Line Number after a set wait time.

Example:

Assume the assigned hotline for 2S2O line 3 is 701 and the Warm Line(Hot Line Delay) is 5 seconds. If no extension number is dialed within 5 seconds, the call will be automatically diverted to the assigned hotline (ext 701). The system allows users to record a voice prompt (e.g. “please enter an extension number or wait for the operator to connect you”) to use in this situation.
Assume the assigned hotline for 2S2O line 4 is 702 and the wait time is 0 second. When P4 receives a call from an outside line, it will be automatically diverted to extension 702.

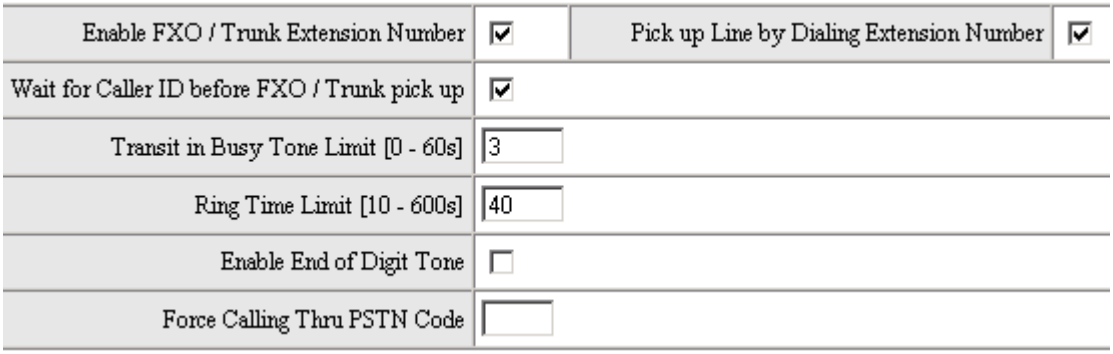
- Dail-out Prefix: It is the number dialed automatically by the system when the FXO interface diverts a call to the PSTN by VoIP. (S series gateways do not have this function).

Example:

If PBX extension needs to dial “0” to make a PSTN call, and the FXO are connected to PBX extension. In this case, the Dial-out prefix should be set to “0”. If the PBX requires some delay time before capturing a line, then the trunk prefix should be set as “0,” so that after dialing a 0, it will pause for 1 second before dialing the destination number. Each comma represents a 1 second delay. If more delay time is required, simply add more commas. Please note that if a Dial-out prefix is set, the line won’t be able to dial to any PBX extension line (FXS interface does not have a trunk prefix function). Please refer to Section 6 if required to dial to a PBX extension line and PSTN concurrently.

- Group Hunting: Select group hunting when there is an incoming call, the gateway will automatically assign an unassigned call according to Hunting Priority. If Line 2 does not want to be set as an assigned line to receive any inbound calls, the function can be disabled. Users can also use the Up or Down key to adjust hunting priority (No setting is required for the FXO interface).

- Grounding Compensation: If the FXO can't work correctly, you will hear noises or echoing in the background as you are talking.
- Enable FAX: Enable this line to detect if there is a FAX tone to transfer the codec.


Enable FXO / Trunk Extension Number
Pick up Line by Dialing Extension Number
Wait for Caller ID before FXO / Trunk pick up
Transit in Busy Tone Limit [0 - 60s]
3
Ring Time Limit [10 - 600s]
40
Enable End of Digit Tone
Force Calling Thru PSTN Code

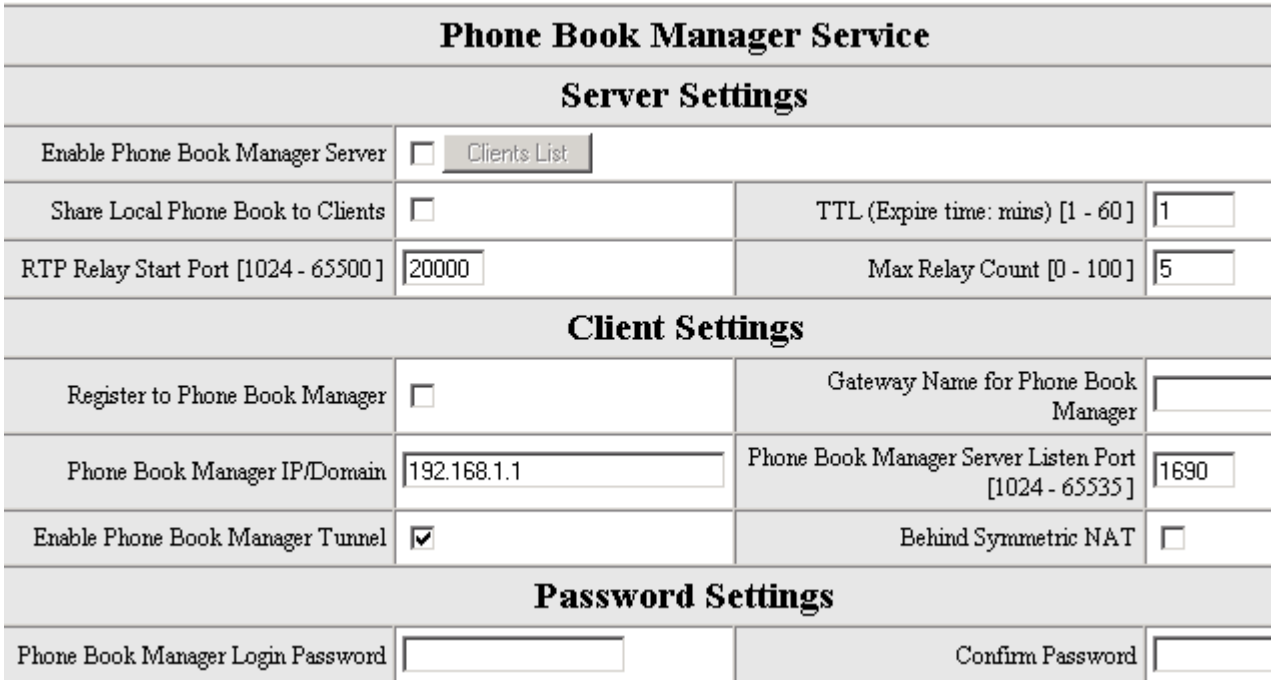
- Enable FXO/Trunk Extension Number: Select this function only when FXO receives 2 or more different PBX or PSTN, or under special circumstances. Users are free to call out from a desired channel, if assigned. If you register to a Proxy it MUST be checked.

Description:
If the user at Line 1 (FXS) of this system wants to assigns Line 4 (FXO) to make a call, he/she can dial 704 22520199.
If this item isn't checked, the gateway will select a line automatically to call out from. For example, dial 22520199 without adding extension number 703 or 704 of the FXO port.

- Pick up Line by dialing extension number: Allows user to dial just the FXO extension - 703 - to use when the PSTN line is connected on the FXO port. If you are registered to a Proxy, it MUST be checked.
- Wait for Caller ID before FXO / Trunk pick up: To detect caller ID from FXO port.
- Transit in Busy Tone Limit: The duration VoIP gateway plays a busy tone before FXO hook-on. To notify the caller from PSTN that this call is finished.
- Ring Time Limit(10 - 600secs)：The timeout to cancel a call when no one answers.
- Enable End of Digit Tone：The VoIP gateway will play a "Beep-Beep" tone to notify the call is in progress.
- Force Calling Thru PSTN code: Dial the code to get a PSTN line for dial out. For example: If you would like dial "23456789" through PSTN, just dial "*33 23456789"

Private Network

Users can establish a private network by Phone Book Manager Service. Phone Book Manager Service is different from Proxy. Gateway is able to register to Phone Book Manager Service and SIP Proxy at the same time.


Phone Book Manager Service
Server Settings
Enable Phone Book Manager Server
Clients List
Share Local Phone Book to Clients
TTL (Expire time: mins) [1 - 60]
1
RTP Relay Start Port [1024 - 65500]
20000
Max Relay Count [0 - 100]
5
Client Settings
Register to Phone Book Manager
Gateway Name for Phone Book Manager
Phone Book Manager IP/Domain
192.168.1.1
Phone Book Manager Server Listen Port [1024 - 65535]
1690
Enable Phone Book Manager Tunnel
Behind Symmetric NAT
Password Settings
Phone Book Manager Login Password
Confirm Password

Server Settings

- Enable Phone Book Manager Server: It allows other Gateway users to register the IP address and Gateway Number in this Phone book manager server. It is recommended to use static IP for the Phone Book Manager Server.
- Share Local Phone Book to clients: While this option is enabled and the gateway is performing as a Phone Book Manager Server, this gateway will append its local Phone Book entries to the Server for other clients to lookup.
- TTL (Time to Live): If a Gateway system that is controlled by the Phone Book Manager Server does not report back within the deadline set by TTL, the system will be excluded from the user's list. Each Gateway should report to the Phone Book Manager Server once every 40 seconds.
- RTP Relay Start Port: Start port of RTP Relay Service. Phone Book Manager supplies RTP Relay to ensure that each client is able to communicate normally.
- Max Relay Count: The max relay service counts supported by Phone Book Manager. If the quality of voice is bad, please decrease Max Relay Count.

Note: Please considerate the bandwidth of Phone Book Manager and the Codec of each client. For example: a client uses Codec as 729A, Packet Time is 20ms, and each call will consume 60kb/s of bandwidth in Phone Book Manager. If the bandwidth is limitless, Every Phone Book Manager is able to support 4 calls and 5 RTP Relay service at the same time.

The bandwidth of different Codec needing:

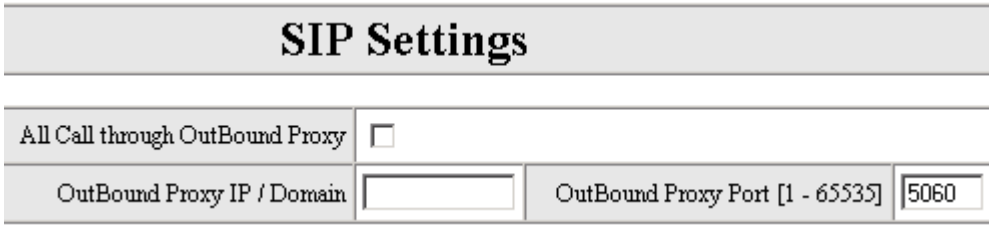
	G.711	G.729a	G.723.1
20ms	171.2	59.2	N/A
30ms	156.8	44.8	41.6
60ms	142.4	30.4	27.2

Client Settings

- Register to Phone Book Manager: To register to the Phone Book Manager. If the Gateway is Phone Manager Server, it has to enable this function to communicate with other clients.
- Gateway Name for Phone Book Manager: The alias registered with the Phone Book Manager Server.
- Phone Books Manager IP/Domain: Enter the IP address for the Phone Book Manager Server. It supports URL (Uniform Resource Locator).
- Phone Book Manager Server Listen Port: The protocol communication port for transmitting signals between the Phone Books Manager and other Gateway systems. Please confirm whether the setting is the same as that for the **Phone Book Manager Server.**
- Enable Phone Book Manager Tunnel: Check it to make a call through private tunnel.
- Behind Symmetric NAT: Check it if the VoIP gateway is built under NAT.
- Phone Book Manager Login Password: Enter the registered password. If this system is serving as the Phone Books Manager, the set password is also the password used for registering other Gateway systems.

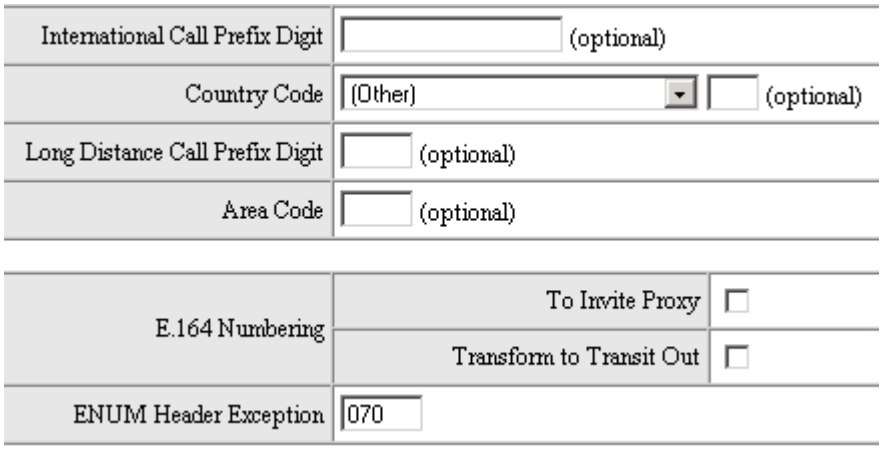
Note: If the NAT type of your router is Restricted Cone, Port Restricted Cone and Symmetric, please use STUN to decrease the loading of Phone Book Manager, and the group is able to loading more calls.

SIP Settings

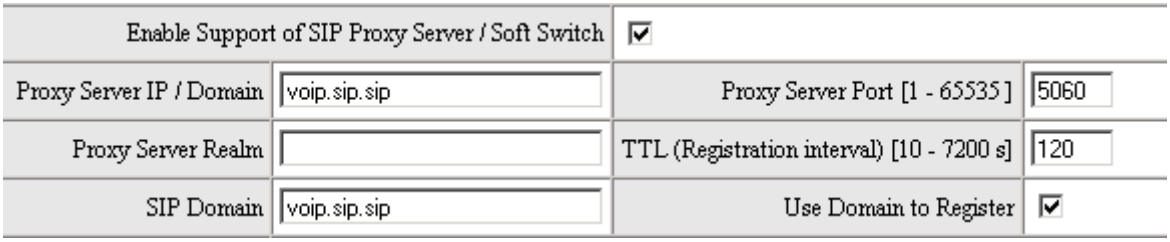


- All Call through OutBound Proxy：An outbound proxy server handles SIP call signaling as a standard SIP proxy server would. Furthermore, it receives and transmits phone conversation traffic (media) in between two talking gateways. This option tells the gateway to send and receive all SIP packets to the destined outbound proxy server rather than the remote gateway. This helps VoIP calls to pass through any NAT protected network without additional settings or techniques. Please make sure your VoIP service provider supports outbound proxy services before enable it.

E.164



- International Call Prefix Digit: Enter the International call prefix.
- Country Code: Users please select the desired country code.
- Long Distance Call prefix Digit: The long-distance prefix digit for making a long-distance call.
- Area Code: Please enter the area code.
- E.164 Numbering: To invite Proxy to follow the E.164 rule. It depends on the Proxy. **If you fail to make a call, please contact your ITSP**.



- Enable Support of SIP Proxy Server / Soft Switch: Enable the functions to inter-work with Proxy Server / Soft Switch.

- Proxy Server IP/Domain: Enter the Proxy Server IP address or URL (Uniform Resource Locator). You can set 3 redundant Proxy spread by semicolon.

  EX: 61.123.231.1;12.34.56.78;proxy.sip.sip

- Proxy Server Port: Enter the Proxy Server **listen** port number. (The factory default value is 5060)
- Proxy Server Realm: Enter the correct registered Proxy Server Realm name to avoid registration failure. **If you fail to make a call, please contact your ITSP.**
- TTL: Enter the desired time interval at which the gateway will report to you Proxy Server.
- SIP Domain/Use Domain to Register: Enter the correct SIP domain to avoid registration failure (it is not necessary to set with some Proxy Servers). If you enable "Uses Domain to Register" the VoIP gateway will register to Proxy with the domain name you filed. Else, the VoIP gateway will register to a Proxy with the IP it resolves. **If you fail to make a call, please contact your ITSP**.

FXS/ FXO Representative number registers to Proxy:

Line	Type	Number	Register	Invite with ID / Account	User ID / Account	Password
Gateway and SIP Prefix Number		11193046	☑		123	123
FXO Representative Number		930467	☑		123	123

Assuming that your registered ID and password are individual, the settings should be as above:

- FXS Representative Number: To register all FXS ports as a hunting group.
- FXO Representative Number: To register all FXO ports as a hunting group.
  All the grouped FXO ports will be hunted automatically.

Each line registers to Proxy independently:

Line	Type	Number	Register	Invite with ID / Account	User ID / Account	Password
Line 1	FXS	11193046 701 Auto	☑	☑	123	123
Line 2	FXO	11193046 702	☑	☑	123	123

- Invite with ID / Account: VoIP gateway can be invited to a VoIP trunk gateway w/o register to a Proxy. Please contact your ITSP

As there are various Proxy Server providers, our company has designed the gateway to be compatible with them, and according to RFC standards. If any registration problem occurs, please consult your Proxy Server provider.

NOTE: When you register with a Proxy Server, dialing principles may vary with different Proxy Servers, especially when dialing through a remote end FXO port.
Please consult your Proxy Server Provider.

Calling Features

Calling Features

Line	Type	Unconditional Forward	Busy Forward	No Answer Forward	Forwarding Number	Call Hold	Call Transfer	Call Waiting	Three-Way Calling / Service ID
FXS Representative Number		☐	☑	(N/A)	2252	(N/A)	(N/A)	(N/A)	
Line 1	FXS	☑	☐	☐ Forward after[10 - 60] 20 s	2253	☑	☑	☑	☑ 996688
Line 2	FXS	☐	☐	☑ Forward after[10 - 60] 20 s	2254	☑	☑	☑	☑ 996688

- Unconditional Forward: All incoming calls will be forwarded to the "Forwarding Number" automatically.
- Busy Forward: Forward the incoming call to "Forwarding Number" when the port is busy.
- No Answer Forward: Forward the incoming call to "Forwarding Number" after ring timeout expires without answer.
- Call Hold: Enable the call hold feature on the specific FXS port.
- Call Transfer: Enable the call transfer feature on the specific FXS port.
- Call Waiting: Enable the call waiting feature on the specific FXS port.
- Three-Way Calling / Service ID: Feature code of conference call defined on Nortel Soft Switch

Calling Feature Instructions:

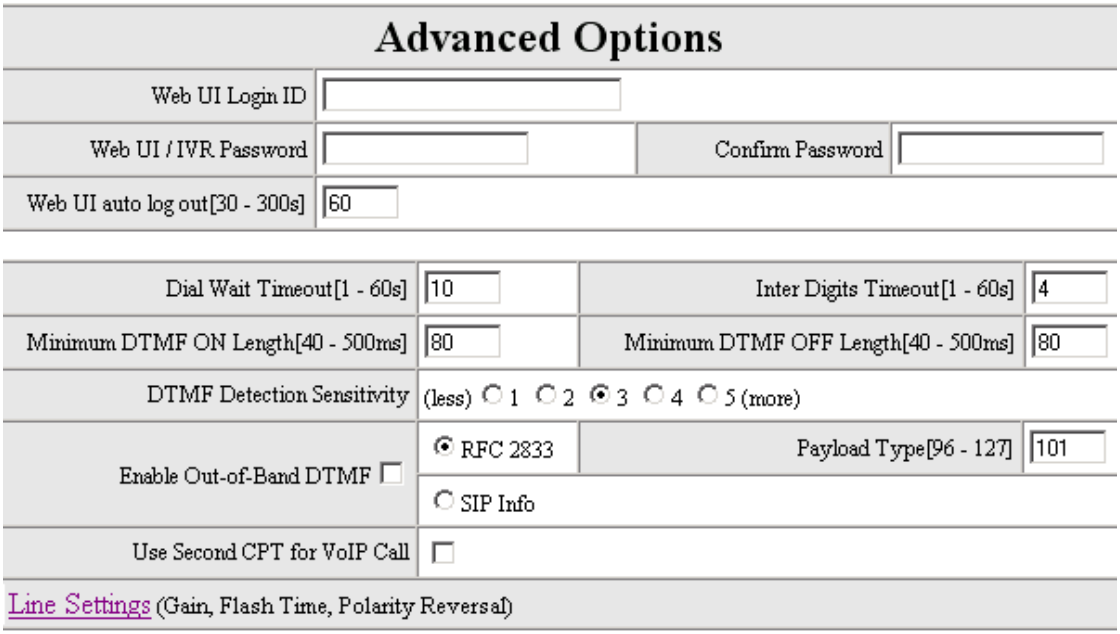
- Call Hold: Ongoing call will be put on hold after FLASH button pressed on the phone set. The gateway will play a repeating music to the remote end.
- Call Transfer: Ongoing call will be put on hold after FLASH button pressed on local phone set (gateway plays a repeating music to the remote end). Meanwhile, local user can dial out to another number after dial tone observed. After the handset is back on the hook, the call on hold will then be transferred to the new call regardless of the status of the new cal. If wrong number is dialed for the new call, just press the FLASH button to get back the call on hold. In another case, if the local user doesn't hang up the phone after new call sets up, press FLASH button to switch between the first call and the new call. Please be informed that PBX between phone sets and VoIP gateway must support FLASH features to make this function work correctly. If a phone set is connecting directly to the FXS port of the gateway and not functioning to FLASH, please adjust the settings in "Flash Detect Time" in category "Advanced Options".
- Example of a Three-Way calling:

  1. Alex dials to Bob, Bob answers that call.
  2. Alex presses Flash and call to Coral (Bob is on hold), Coral answers that call.
  3. Alex dials *61 then presses Flash, thus conference call is created.

  Or

  1. Alex dials to Bob, Bob answers that call.
  2. Coral dials to Alex (Call Waiting), Alex presses Flash to pick the second call and talk to Coral.
  3. Alex dials *61 then presses Flash, thus conference call is created.

Advanced Options



- Web UI Login ID and Password: Enter login ID and password when you log onto the Web.
- Web UI auto log out: If a user does not act within the effective time range when logging into a web page, the user will be disconnected from the web page to allow others to login.
- Dial Wait Timeout: Use it to set the waiting time for the user's first key pressing when dialing a number. The user will hear a busy tone if he/she does not press the first key within the set time frame.
- Inter Digits Timeout: Set the waiting time between each key pressing. The inputted numbers will be dialed after the timeout.
- Minimum DTMF ON Length (Dial on)/ Minimum DTMF OFF Length (Dial off - between tones): Used to set dial tone when a call is being diverted to another extension.
- DTMF Detection Sensitivity: Used to adjust the sensitivity of the telephone keys.
- Enable Out-of-Band DTMF: To send DTMF keys (0~9, *, #,) follow the RFC2833 rules or via SIP Info.
- Payload Type：Payload type of RFC2833.
- Uses Second CPT for VoIP Call: This function is usually applied when the user selects VoIP as the primary path for outgoing calls and PSTN as the backup. By enabling this function, the gateway will generate a different set of tones to inform the user that VoIP is in service. Should VoIP fails and fallback to PSTN, the user will hear PSTN tones instead of the second set CPT. (for CPT related settings, please refer to Trunk Management -> CPT Settings)

Line Settings

Line Settings								
	Extension Number	Type	Listening Volume	Speaking Volume	Tone Volume	Flash Time	Enable Polarity Reversal	PSTN Answer Detection
Line 1	9901	FXS	0 All	0 All	5 All	0.6 All	☐	
Line 2	702	FXO	0	0	5	0.6	☐	Disable

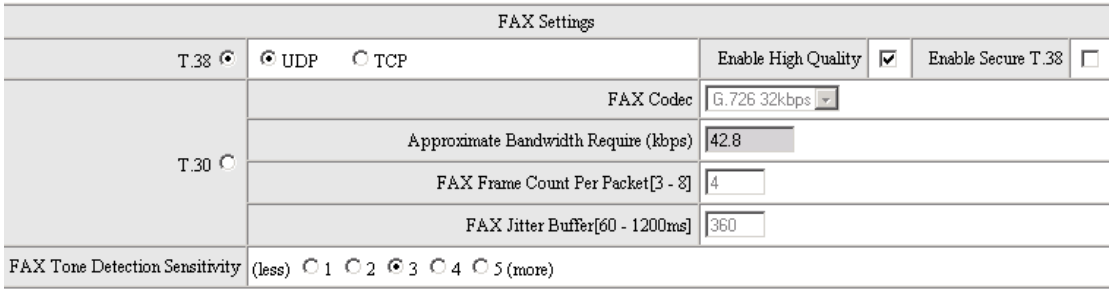
- Listening Volume: Adjusts the hearing volume.
- Speaking Volume: Adjusts the speaking volume.
- Tone Volume: Adds a new option to make tone volume adjustable. This setting will be applied to all tones generated by the VoIP gateway including Dial Tone, Busy Tone, and so on.
- Flash Time:
  FXS: Used to adjust the detecting period of flash signal from the phone set connected to the FXS port. For example, if pressing the HOLD key will disconnect a call, increase the "Flash Detect Time" should fix this issue.
  FXO: Used to set the time frame that FXO generates a FLASH signal.
- Enable Polarity Reversal:
  FXS: As the remote site answer this call or hook on the FXS port will reverse the polarity.
  FXO: This option forces the gateway to detect the reversal of polarity on FXO port as the primary signal to drop a call. Some telephone switches or PBX reverse the line polarity to inform the remote end to drop an ongoing call. Please consult with the telephone service provider for availability of this feature.
- PSTN Answer Detection: This is used only by ITSP.

Voice

Voice				
Preferred Codec Type	G.723.1 6.3kbps			
Jitter Buffer[60 - 1200ms]	360			
Silence Suppression	☐	Echo Cancelling	☑	
Codec	G.711	G.723	G.726	G.729A
Packet Time (ms)	60	60	60	60
Approximate Bandwidth Require (kbps)	71.2	13.6	39.2	15.2

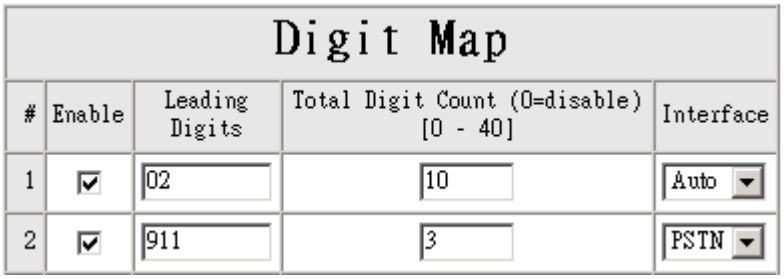
- Preferred Codec Type: Since different voice codec have different compression ratios, so the sound quality and occupied bandwidths are also different. It is recommended to use the default provided (G.723.1) because it occupies less bandwidth and will provide better sound quality.
- Jitter Buffer: Adjusts the jitter to receive a packet. If the jitter range is too wide, it will delay voice transmission.
- Silence Suppression: If one side of a connection is not speaking, the system will stop sending voice data (package) to decrease bandwidth usage.
- Echo Canceling: Prevents poor telecommunication quality caused by echo interference.
- Packet Time: Defines how long the VoIP gateway sends a RTP packet-voice packet- to the other side. The smaller the value, the more bandwidth usage. The larger the value, the more voice delay.
- Approximate Bandwidth Require: The bandwidth required varies with codec format and packet time.

Fax Settings



- T.38: The T.38 protocol is used for better and faster facsimile transmission. When this function is enabled, the following fax and voice parameter settings will be disabled, so it is recommended to enable this function to gain better fax quality. When this function is enabled, please select UDP, TCP, or AUTO. If selecting TCP and some routers cannot use the Fax function, please select UDP instead.
- Enable High Quality: The system sends the same FAX frame twice to get a high quality of the FAX. It requires more bandwidth.
- Enable Secure T.38: This allows VoIP gateways to send faxes on both sides.
- T.30: The system uses T.30 as the protocol for fax transmission. The parameter settings are the same as for voice transmission. However, enabling the fax function will consume more network resources and will affect transmission quality.
- FAX Detect sensitivity：Used to adjust the sensitivity of detection as to whether a phone call is a FAX or not.

Digit Map

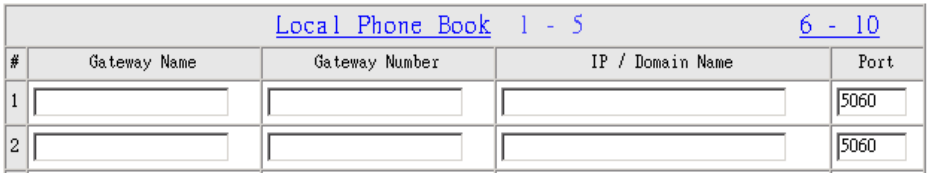


There are 50 sets of leading digit entries to choose voice routing interface – Auto select, PSTN or VoIP.

- Enable: Enable detection of this entry.
- Leading Digits: The leading digits for gateway to scan while user is dialing.
- Total Digit Count: Total number of digits that gateway should accept. 0 disables this feature.
- Interface: The interface calls should go through if above conditions satisfied.

Local Phone Book

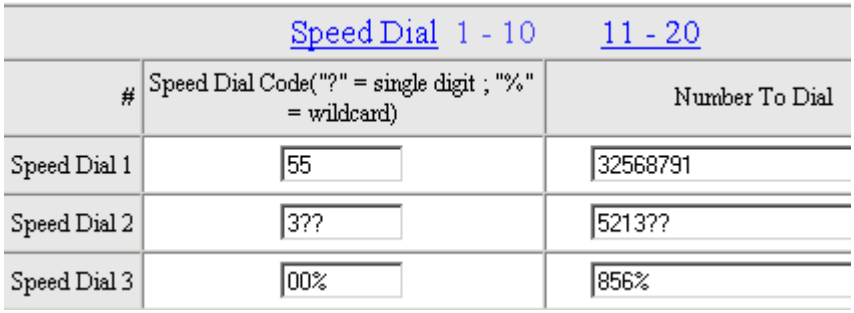
This system can set up and store 100 phone numbers into a phone book and provides an IP address query when calling to other Gateway(s). If no Phone books manager is set within a Gateway group, then all Gateway systems have to set up phone data for each VoIP gateway to communicate with each other.



- Gateway Name: Enter Gateway code or an easy-to-remember name.
- Gateway Number: Enter the desired Gateway number.
- IP/Domain Name: Enter the IP address or URL (Uniform Resource Locator) of Gateway.
- Port: Enter the gateway's gateway port.

Speed Dial

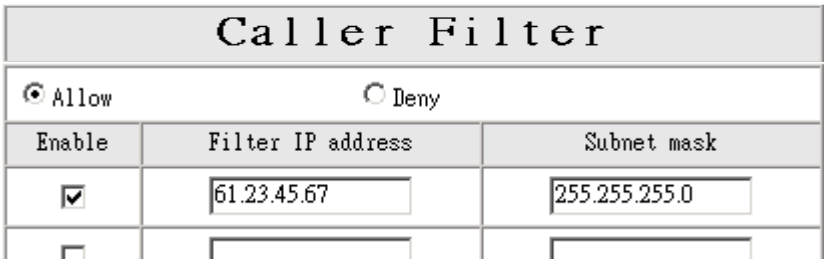
This system can set up 100 numbers for speed dialing. Setting methods are as follows:


Speed Dial 1 - 10 11 - 20
Speed Dial Code("?" = single digit ; "%" = wildcard) Number To Dial
Speed Dial 1 55 32568791
Speed Dial 2 3?? 5213??
Speed Dial 3 00% 856%

Method 1- Single mapping: Fill a short code into the "Speed Dial Code" column, and enter the desired phone number into the "Number To Dial" column.
For example, pick up the handset and dial **55#** and the system will dial 32568791.

Method 2- Multi mapping; Fill the prefix code into the " Speed Dial Code" column and the format to transfer into the "Number To Dial" column.
For example, pick up the handset and dial **301#**, and the system will dial 521301.
If the user dial 00 1657987456321, the system will DIAL 856 1657987456321

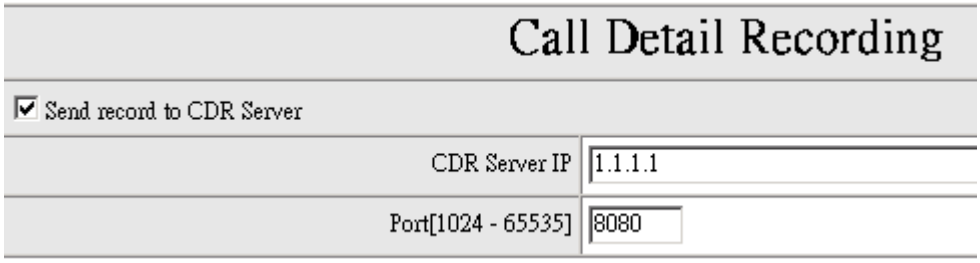
Caller Filter



This function is used at allow or deny SIP Invite from the Proxy list ONLY.

- Filter IP Address: Fill up with the start IP you would like to allow/deny.
- Subnet mask: Fill up with the subnet mask you would like to allow/deny.

CDR Settings



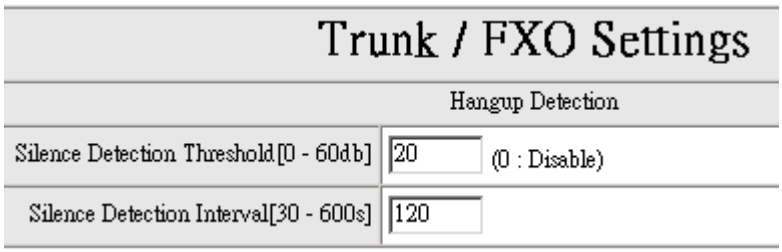
The user can set up a CDR Server to record call details for every phone call.
The present CDR provides the call detail recording in a text file and if needed. it can be imported to prepare for an analysis report.

- Send record to CDR Server: Enables the call detail recording function.
- CDR Server IP: Enter the IP address of the CDR server.
- Port: Enter the listen port of the CDR server.

Language

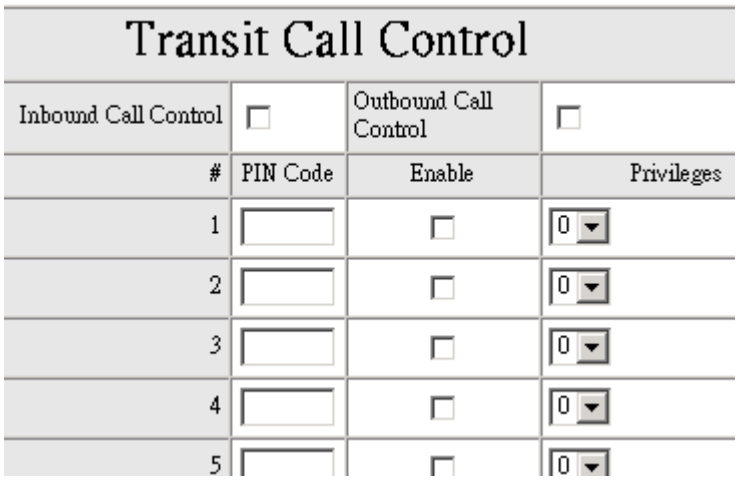
The system provides English, Traditional Chinese, and Simplified Chinese to display text on Web pages. Meanwhile, it will change the language for IVR (Interactive Voice Response).

Trunk /FXO Settings



- Silence Detection Threshold: The volume below the threshold is used as a standard to determine whether or not to hang up the phone.
- Silence Detection Interval: If the detected volume is below the threshold and the time exceeds the silence detection interval, the system will hang up the phone automatically to avoid keeping the line engaged.

Transit Call Control



- Inbound Call Control: To determine when users make a phone call from a PSTN to Gateway FXO whether or not they check the inbound PIN code while using a VoIP －only effective for incoming calls calling from a PSTN trunk.
- Outbound Call Control: To determine when users utilize Gateway FXO interface to divert to a PSTN whether or not they check the outbound PIN code － only effective for outgoing calls being diverted to a PSTN Trunk.
- PIN Code: Enter the PIN code (4-6 digits or leave blank. A blank indicates no PIN code is required at this level. Generally, the PIN at level 5 can remain blank to simplify the phone number.)
- Enable: Enables the PIN code at each level.
- Privileges: The level is divided into 0~5 (The levels are in descending order; 0 stands for the highest authority and 5 stands for the lowest.)

Description: If you wish to restrict a general user (one who is not required to enter the PIN code) to local calls only and prohibit him/her from making long-distance calls started with a prefix "0", do the following steps: 1. Enable the Outbound Call Control function, 2. Set the PIN code for Outbound Level 5 to blank, 3. Set the Long-Distance Control Table to correspond with the Outbound Level 5 to prohibit making any call with the prefix "0" (as shown above).

Dialing principle to enable PIN code

***Inbound Call Control PIN Code * Outbound Call Control PIN Code * Phone number**
Using * to the separate PIN code and the phone number based on actual settings. The PIN can also be omitted.

Actual operation is shown below:

If making a call from Gateway A to Gateway B, and diverting to PSTN (local call)
When Gateway A and Gateway B has both disabled the Outbound Call Control, dial as follows:

Gateway B Number + Local Phone Number
For example: 5168 35020311#

When Gateway A disables Inbound Call Control and Gateway B enables Outbound Call Control, then dial as follows:

*** Gateway B PIN Code * Gateway B Number Local Phone Number**
For example: *123456* 5168 35020311# (Gateway B PIN Code: 123456**)**

When Gateway A enables Inbound Call Control and Gateway B disables Outbound Call Control, then: If using an extension line at Gateway A to make a local call to a remote end through Gateway B, then dial

Gateway B Number + Local Phone Number
For example: 5168 35020311#

If using a trunk line to dial to Gateway A, and then making a local call to a remote end through Gateway B, then dial

*** Gateway A PIN Code * Gateway B Number Local Phone Number**

When the instruction "Enter phone number" is given, then dial
For example: *111111* 5168 35020311# (Gateway A PIN Code: 111111**)**

If using a trunk line to dial to Gateway A and then to the extension line of Gateway B, dial

*** Gateway A PIN Code * Gateway B Number**
When the instruction "Enter phone number" is given.
For example: *111111* 5168#

When Gateway A enables Inbound Call Control and Gateway B enables Outbound Call Control, then If using an extension line at Gateway A to make a phone call to a remote end through Gateway B, then dial

*** Gateway B PIN Code * Gateway B Number Local Phone Number**
For example: *123456* 5168 35020311# (Gateway B PIN Code: 123456**)**

If using a trunk line to dial to Gateway A, and then dialing a local call to a remote site through Gateway B, please dial

*** Gateway A PIN Code * Gateway B PIN Code * Gateway B Number Local Phone Number**
For example: *111111* 123456* 5168 35020311# (Gateway A PIN Code 111111 Gateway B PIN Code 123456)

Long-Distance Control Table

This table controls the level of authority of an outgoing call that is dialed through FXO and diverted to PSTN, as below:

Long Distance Control Table

#	0	1	2	3	4	5
1	0204					
2						
3						
4						
5						

The long-distance control table controls the level of authority of an outgoing call through FXO.
Instruction:

- If Level 0 (the highest level) is set to prohibit dialing any number started with prefix 0204, then any level below 0 (including Levels 1 to 5) is also prohibited.

- If Level 1 is set to prohibit dialing any number with prefix 0, then any level below 1 (including Levels 2 to 5) is also prohibited. Since Level 0 is not restricted to any prefix, therefore at level 0 users can dial a number with the prefix 0.

Principle: Downward Restriction —If the users at a higher level cannot dial a number with a certain prefix, then users at lowers level also cannot dial a number with the same prefix.

Long Distance Exception Table

This table handles any exceptions to the long-distance call table.

Long Distance Exception Table

#	0	1	2	3	4	5
1	020488988					
2						
3						
4						
5						

Instruction:

- According to the Long Distance Control Table, users at Level 2 are prohibited to dial a number with the prefix 0. But, if a prefix 09 is set in the Exception Table, then the users at Levels 1 and 2 could then dial any number with the prefix 09.
- According to the Long Distance Control Table, users at Level 0 are prohibited to dial a number with the prefix 0204. But, if the number 020488988 is set in the Exception Table as above, then users could then dial this number.

Principle: Upward Opening —If the users at a lower level can dial a number with a certain prefix, then the users at higher levels can also dial a number with the same prefix.

CPT/Cadence Settings

CPT/Cadence setting parameters serve as the basis of an FXO interface to determine whether or not a PSTN-call receiving party has hung up the phone. If the following parameters differ from the parameters of the actual assigned lines, it could cause the FXO to continue to engage a line.

Busy Tone Cadence Measurement

Busy Tone Cadence Measurement

	T_ON_1	T_OFF_1	T_ON_2	T_OFF_2	Auto Learning	
BTC # 1	0	0	0	0	☑	200 / 200 Setting
BTC # 2	0	0	0	0	☑	200 / 200 Setting
BTC # 3	0	0	0	0	☑	200 / 200 Setting
BTC # 4	0	0	0	0	☑	200 / 200 Setting
BTC # 5	0	0	0	0	☑	200 / 200 Setting
BTC Detection Sensitivity	(less) ○ 1 ○ 2 ○ 3 ◉ 4 ○ 5 (more)					

- Busy Tone Cadence Measurement: Provide a best solution of FXO integrated with PSTN or PBX. FXO will learn the busy tone automatically.
- BTC Detection Sensitivity: The more sensitivity, the more quickly the system will cut off the call. If the system often cut off an un-finished call, select less sensitivity.

CPT parameters Table

# 1 Enable Detect Setting 1						Default
Tone Type	Low Frequency	High Frequency	T_ON_1	T_OFF_1	T_ON_2	T_OFF_2
Dial Tone	350	440	3000	0	0	0
Congestion Tone	480	620	250	250	0	0
Busy Tone	480	620	500	500	0	0
Ring-Back Tone	440	480	1000	2000	0	0

The CPT has 3 sets of parameter tables. Please adjust the CPT based on local PSTN or PBX.

- Moreover, users can use CPT Auto Detect to detect CPT parameters. Instructions are shown in the following section.
- Using UDT detection: If the CPT auto detect function is not able to determine whether or not a PSTN-call receiving party has hung up the phone, then the UDT detection function can serve as a back up. To do this, enter the high/low frequency parameters from the CPT table into the UDP table.
- The method to detect CPT are as followed:

NOTE: To cope with different local PSTN and different PBX models, the system provides CPT Auto Detect function to prevent the FXO from engaging a line. However, if the line of the receiving party is engaged and his/her PSTN uses a voice prompt to replace the traditional beep sound, then the system would not be able to detect a busy tone. Silence detection should then be used to determine whether or not to end the call.

CPT Auto Detect

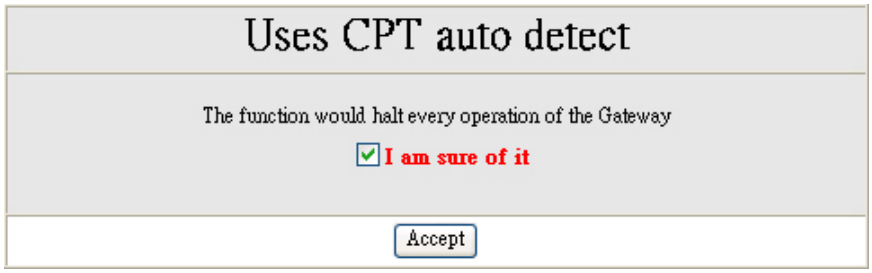
- 2 PSTN phone numbers or 2 PBX extension lines are needed.
- Connect one of the phone sets to the FXO port (P3 or P4 of MODEL 2S2O).
- The line of "Dial Number" must be hook on. Set the outgoing phone number the same as the phone line that is in use as above, and click the "Accept" button to start detection.
- If detection is successful, the parameters obtained will automatically be inserted into the CPT parameter.
- Save settings and restart the system.

Detailed description is given below:

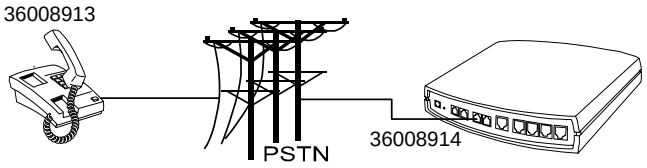
Click "CPT auto detect" at the bottom of Trunk CPT Settings.

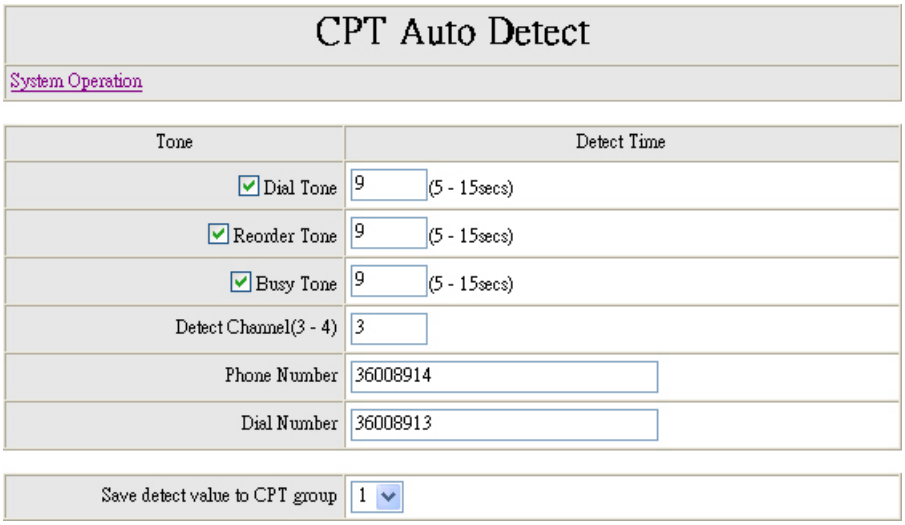
Disable UDT	UDT Setting		# 1 Disable	# 2 Disable	# 3 Disable	# 4 Disable
Reorder Tone / Busy Tone		Low Freq	300	300	300	300
		High Freq	0	0	0	0
Uses CPT auto detect						

If CPT auto detect is used, the function would halt every operation of the Gateway. Select **"I am sure of it"**, and then click the "Accept" button. Wait for 15 seconds, and then you will enter CPT auto detect window.



Direct Connection to PSTN





- Connect one of the trunk lines to the Gateway FXO Port (For MODEL 2S2O, please connect to P3).
- The line of “Dial Number” (36008913) must be hook on.
- Detect Channel: Enter 3 (The trunk line is connected to P3, and uses P3 for outgoing detection).
- Phone Number: Enter the number of the FXO line.
- Dial Number: Enter the number of the end to be tested—36008913.
- CPT grout: Enter the group that after testing to replace with.
- Finally, click the “Accept” button.

Detection in Progress: during detection, the following windows will appear.

Once detection of a busy tone is in progress, the system will dial the number to be tested (in this case

36008913). After it rings pick up the phone and enter"#", then hang up. The system will then detect a busy tone automatically.

Current operation

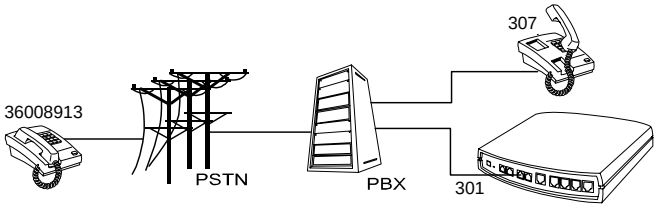
```
Detecting Busy Tone...
Please hook-off after the extension ring up, then press
[#] before hook-off.
```

After detecting it will be as below:

Connected to a PBX Extension Line

If Gateway is connected to a PBX extension line, then the busy tone of both the PBX and the PSTN must be detected.

Connecting to a PBX extension line and detecting the busy tone of the PBX



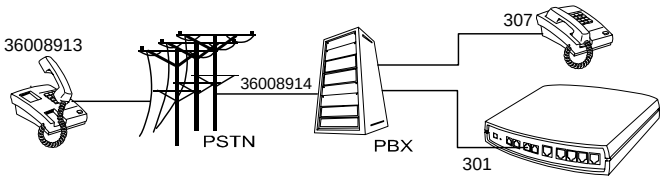
CPT Auto Detect

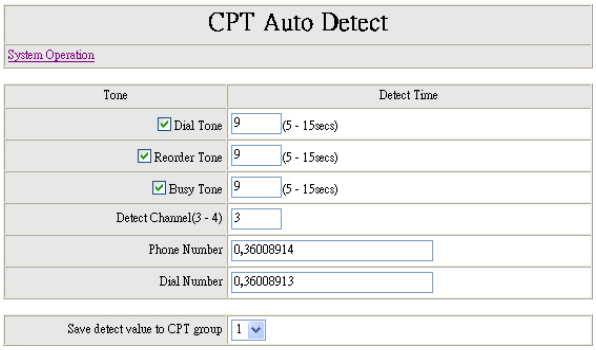
System Operation

Tone	Detect Time
Dial Tone	9 (5 - 15secs)
Reorder Tone	9 (5 - 15secs)
Busy Tone	9 (5 - 15secs)
Detect Channel(3 - 4)	3
Phone Number	301
Dial Number	307
Save detect value to CPT group	1

- Connect one of the PBX extension lines to the Gateway FXO Port (For MODEL 2S2O, connect the line to P3).
- Detect Channel: Enter 3 (The trunk line is connected to P3, and uses P3 for outgoing detection).
- Phone Number: Set the number of FXO line –to detect Reorder Tone.
- Dial Number: Enter the number of the end to be tested—307.
- CPT grout: Enter the group that after testing to replace with
- Finally, click the "Accept" button.

Connected to a PBX extension line to detect the busy tone of PSTN

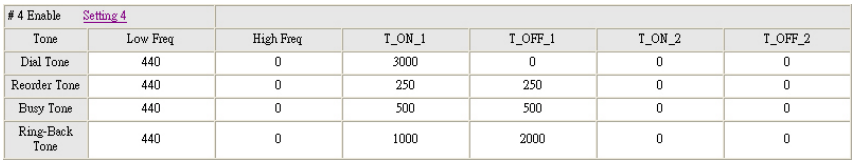




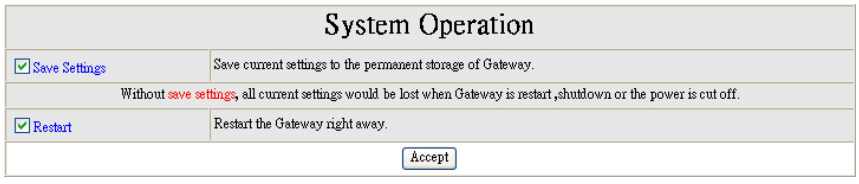
- Connect one of the PBX extension lines to Gateway FXO Port (For MODEL 2S2O, connect the line to P3).
- Detect Channel: Enter 3 (The trunk line is connected to P3, and uses P3 for outgoing detection).
- Phone Number: Set the number of FXO line –to detect Reorder Tone.
- Dial Number: Enter the trunk prefix of a PBX, comma, and the number of the end to be tested -- 0,36008913. (Assuming that an extension line is set to dial 0 to make outgoing calls).
- CPT grout: Enter the group that after testing to replace with. If connecting Gateway to a PBX extension line, please do not set the detected busy tone of the PBX and the PSTN in the same set, otherwise the value detected the first time will be overwritten.
- Finally, click the "Accept" button.

Filling in the CPT Table

Fill in the table after the detection is completed as below, where the values are the frequency and On-and-Off ratio detected. Please click the "Accept" button. If connecting Gateway to a PBX extension line, please do not set the detected busy tone of the PBX and the PSTN in the same set, otherwise the value detected the first time will be overwritten.



Save Settings


System Operation
Save Settings
Save current settings to the permanent storage of Gateway.
Without save settings, all current settings would be lost when Gateway is restart ,shutdown or the power is cut off.
Restart
Restart the Gateway right away.
Accept

Save the settings and restart the system after the test is completed. Then click the "Accept" button. Gateway will use the new parameter to detect whether a call has ended.

System Information

System Information

Port Status

No	Type	Extension Number	Line Status	Calls	Dialed Number	Proxy Register	UPnP on RTP
1	FXS	701		0		Disabled	
2	FXS	702		0		Disabled	
3	FXO	703		0		Disabled	
4	FXO	704		0		Disabled	

Server Registration Status

DDNS Registation	Disabled
Phone Book Manager Registation	Disabled
STUN Registation	Disabled
UPnP Negotiation	Disabled

RTP Packet Summary

Displays the information of the last finished call. It contains peer IP, peer port, packets sent, packet received and packet lost.

RTP Packet Monitor

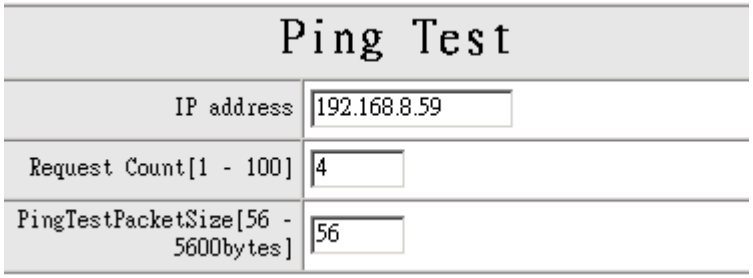
Line 1	G.711 64kbps	Send Packet Count	0	Receive Packet Count	0	Packet Lost Count	N/A
The last packet's source IP				The last packet's source Port		0	
Line 2	G.711 64kbps	Send Packet Count	0	Receive Packet Count	0	Packet Lost Count	N/A
The last packet's source IP				The last packet's source Port		0	
Line 3	G.711 64kbps	Send Packet Count	0	Receive Packet Count	0	Packet Lost Count	0

STUN Inquiry

STUN Inquiry

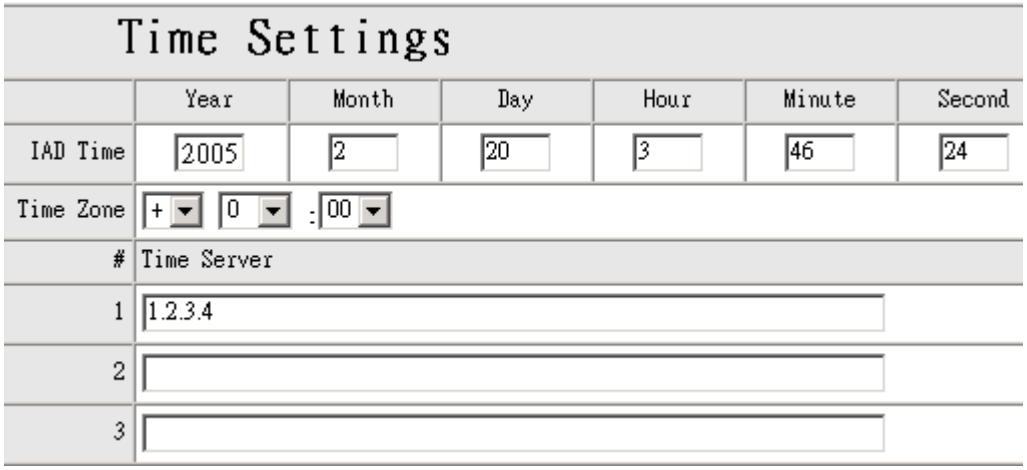
NAT Type	Open
STUN Server	1.2.3.4
STUN Server Port(1024 - 65535)	3478

Ping Test



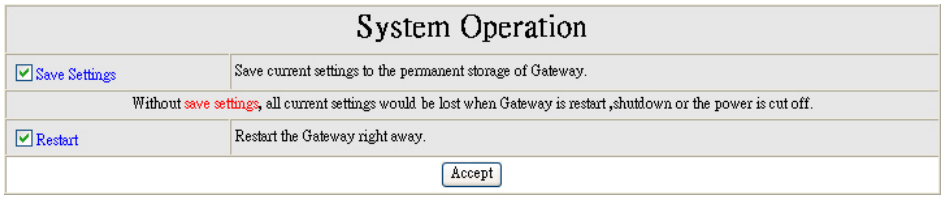
Use “ping” to identify if the remote peer is reachable. Fill in remote IP address and click “Test” will start the test.

Time Settings



Time Zone: Set the Time Zone where VoIP gateway resides.
Time Server #1~#3: Set the Time Server where VoIP gateway should sync up during start up. (NTP protocol)

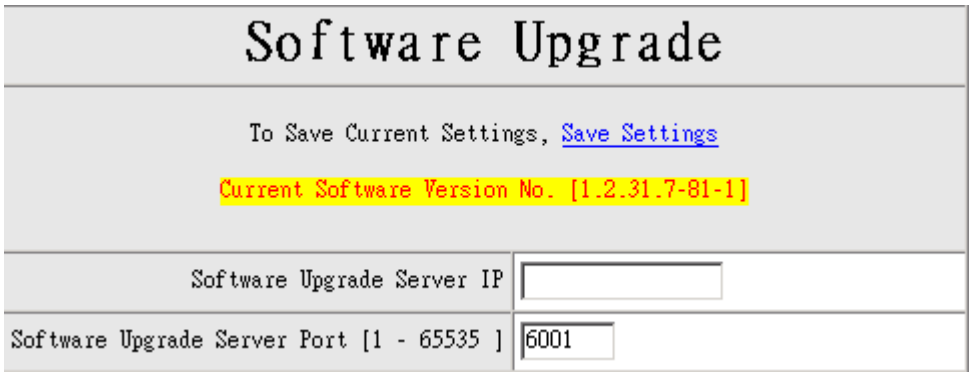
System Operations (Save Settings)


System Operation
Save Settings
Save current settings to the permanent storage of Gateway.
Without save settings, all current settings would be lost when Gateway is restart ,shutdown or the power is cut off.
Restart
Restart the Gateway right away.
Accept

- Save Settings: Save settings after completing. The new settings will take effect after the system is restarted. Please select “Save Settings”.
- Restart: If it is necessary to restart the system, please select “Restart” and click the “Accept” button.

Software Upgrade

Gateway provides software upgrade function for a remote end.


Software Upgrade
To Save Current Settings, Save Settings
Current Software Version No. [1.2.31.7-81-1]
Software Upgrade Server IP
Software Upgrade Server Port [1 - 65535]
6001

- Software Upgrade Server IP address: Please enter the software server IP address.
- Software Upgrade Server Port: The default setting is 6001(Do not change this setting).

Logout

Gateway only allows one user to login at a time, so whenever a change is made, please save the settings, restart the system, or logout to avoid the situation where other users cannot login to change settings.

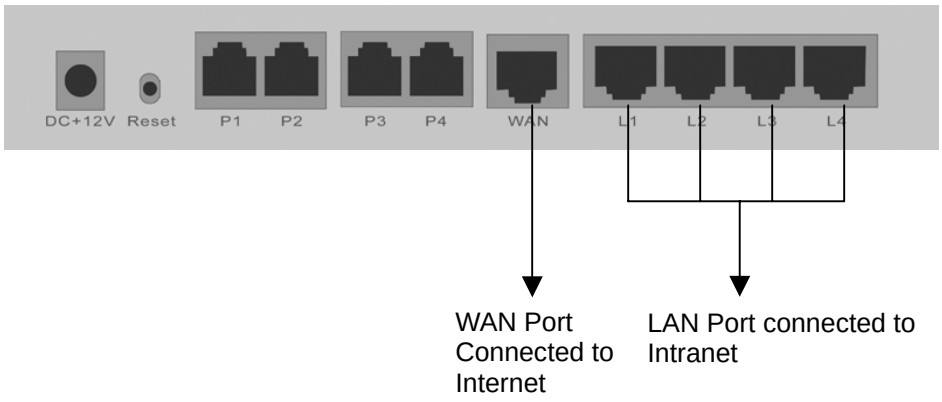
5. IP Sharing Functions

All Gateway series have a built-in IP sharing function. The settings and instructions at a PC end are described below:

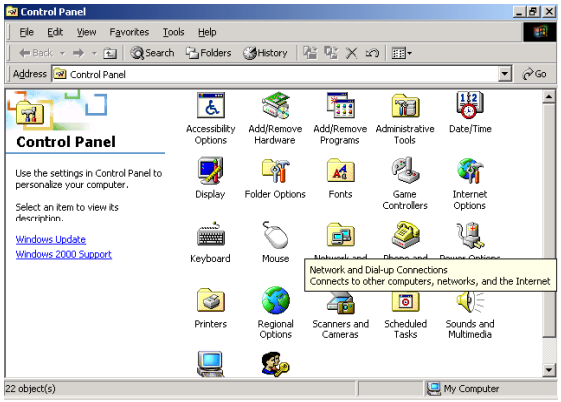
Current Intranet only supports static IP mode, and the settings at the PC end are as follow:
Available IP address Range : 192.168.8.1 – 192.168.8.253 (default address of Gateway is 192.168.8.254)
Subnet Mask : 255.255.255.0
Default Gateway : 192.168.8.254

The above values vary with different LAN Port Settings.
Assume Gateway's LAN settings are,
IP address : 192.168.3.1
Subnet Mask : 255.255.255.0

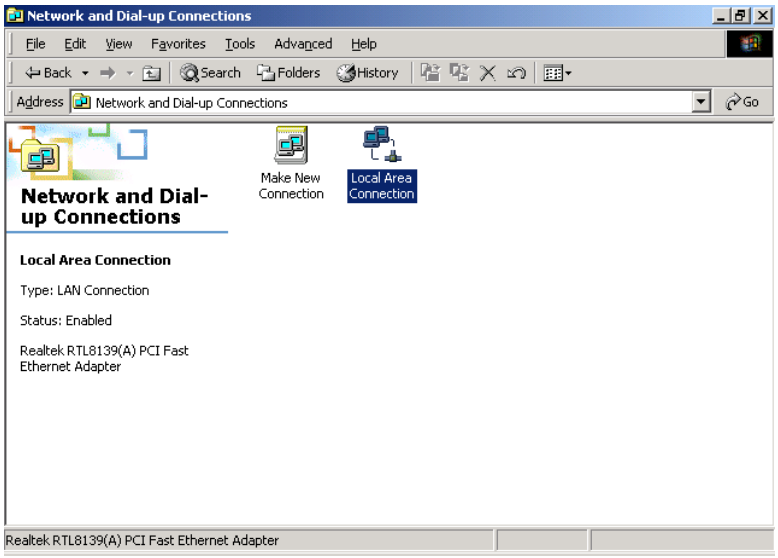
Then, the settings at PC end should be as follows:
Valid IP address range: 192.168.3.2 – 192.168.3.254
Subnet Mask : 255.255.255.0
Default Gateway : 192.168.3.1



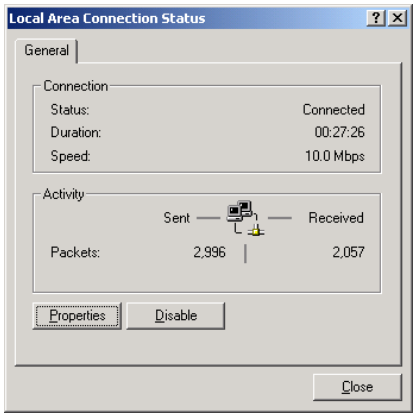
The IP settings on PC are as follows (using Windows 2000 for example)
Open Start->Settings->**Control Panel**
Open **Network and Dial-up Connection**



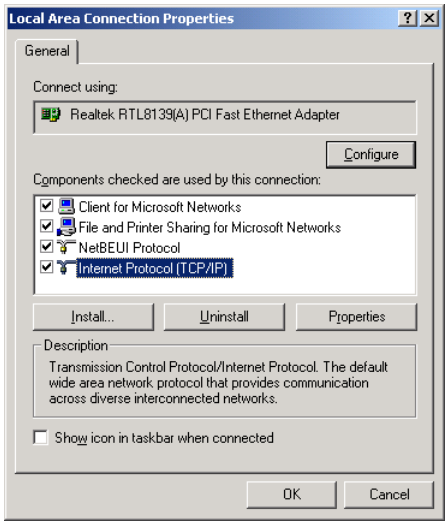
Open Local Area Connection



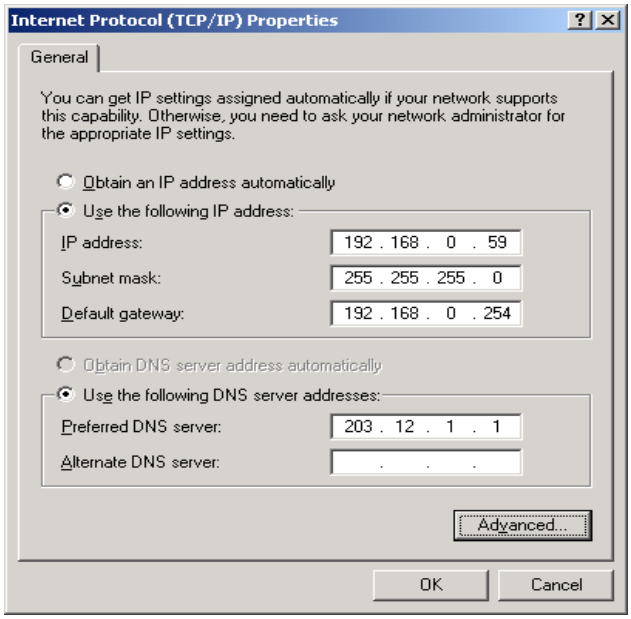
Click **Properties**



Select **TCP/IP**, and then click **Properties**.



Select "**Use the following IP Address ”** and enter IP address, Subnet Mask, and Default Gateway. Please note that an IP address in the same domain cannot be reused. Then, enter the DNS server IP address (varies in different networks. consult your ISP's service for information). Click the "**OK”** button and after completing the settings, users can use both the VoIP and network services concurrently.


Internet Protocol (TCP/IP) Properties
General
You can get IP settings assigned automatically if your network supports this capability. Otherwise, you need to ask your network administrator for the appropriate IP settings.
Obtain an IP address automatically
Use the following IP address:
IP address: 192 . 168 . 0 . 59
Subnet mask: 255 . 255 . 255 . 0
Default gateway: 192 . 168 . 0 . 254
Obtain DNS server address automatically
Use the following DNS server addresses:
Preferred DNS server: 203 . 12 . 1 . 1
Alternate DNS server:
Advanced...
OK
Cancel

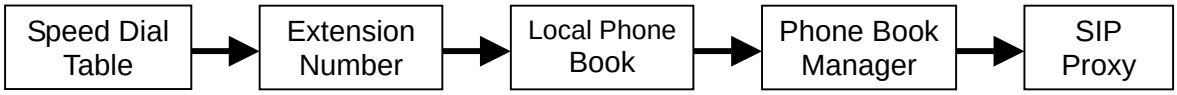
6. Coding Principle

Instruction

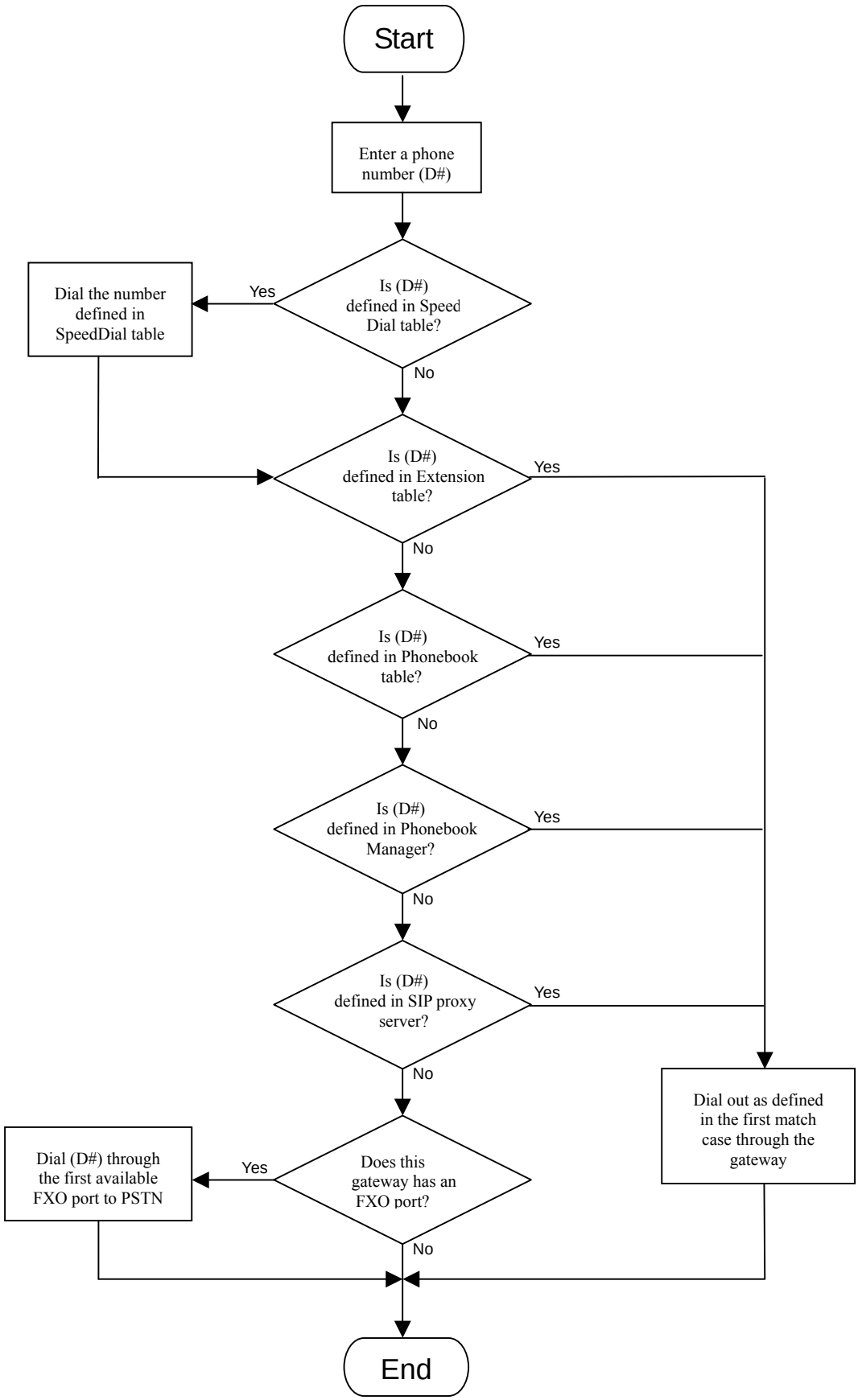
- After a phone number is entered, dial # to call out immediately or, wait until the "Inter DTMF Timeout" expires (defined in "Advanced Options", default=4 seconds).
- If the phone number fits the setting of Digit Map, the gateway dials out the phone number through the assigned interface automatically.
- The phone number should have at least 2 digits (not including * and #).

Dialed Number Processing Flow

To maintain maximum flexibility, the number dialed will be looked up from several tables defined in the gateway. Once no match can be found, it will look up again from the registered SIP Proxy Server. The look up flow is shown below:

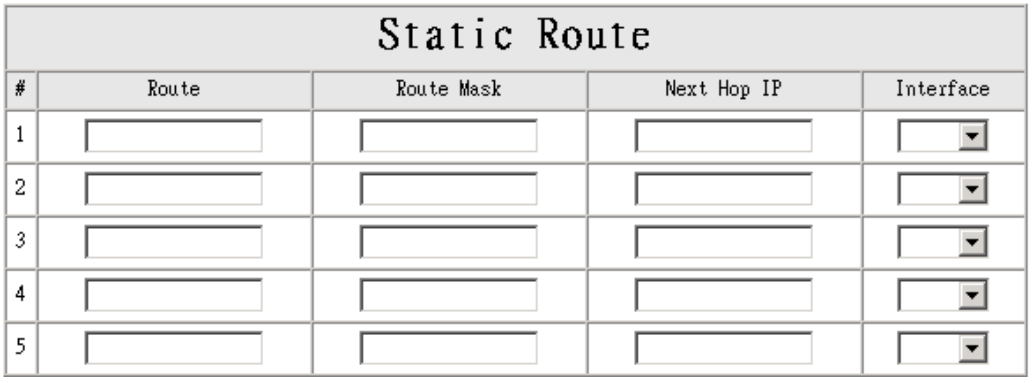


A complete flow chart is on the next page.


Start
Enter a phone number (D#)
Is (D#) defined in Speed Dial table?
Yes
Dial the number defined in SpeedDial table
No
Is (D#) defined in Extension table?
Yes
No
Is (D#) defined in Phonebook table?
Yes
No
Is (D#) defined in Phonebook Manager?
Yes
No
Is (D#) defined in SIP proxy server?
Yes
No
Does this gateway has an FXO port?
Yes
Dial (D#) through the first available FXO port to PSTN
No
Dial out as defined in the first match case through the gateway
End

7. Advanced Feature

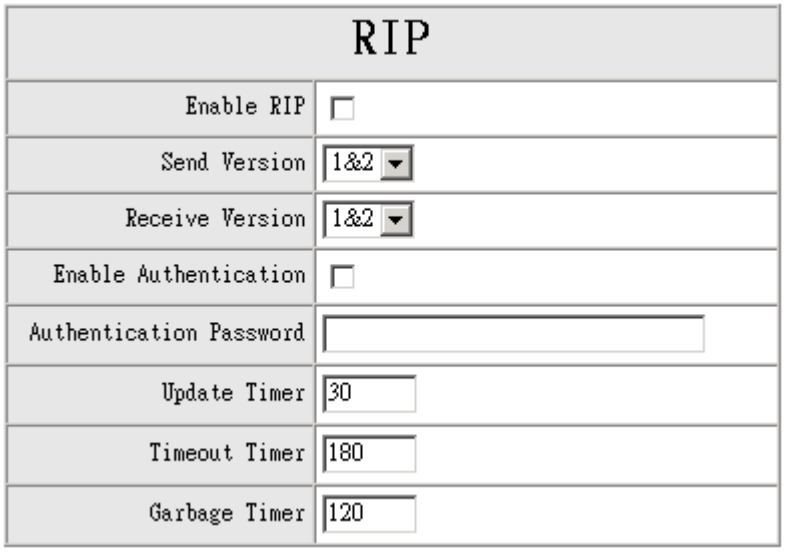
Static Route※



To build static routes within an internal network. These routes will not apply to the Internet.

- Route: Enter the IP of the specified network.
- Route Mask: Enter the subnet mask to be used for the specified network.
- Next Hop IP: Enter the IP address to the specified network.
- Inter Face: Select the inter face.

RIP(Routing Information Protocol) ※

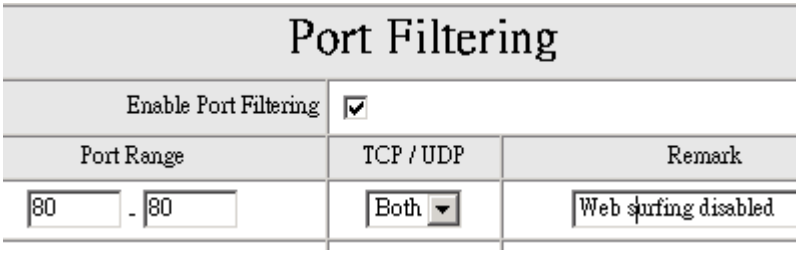


To establish dynamic routes within an internal network. These routes will not apply to the Internet.

- Enable Authentication/Authentication Password: If the box is checked. All the boxes in this RIP group should be filled in the same password.

Port filtering

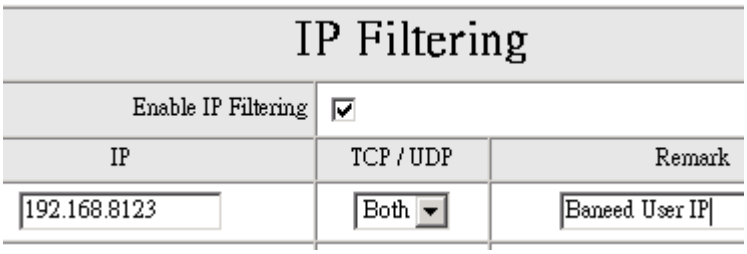
Port filtering enables you to control all data that can be transmitted in routers; **principles of filtering---When the port used at the source end is within the limited scope, it will be filtered without transmission.**



- Enable port filtering: whether to enable this function or not.
- Port Range: Set the range of port to be filtered, suppose it is 80 and when use protocol is Both or TCP, all computers will be unable to use the services of http (port 80)— will be unable to browse normal WebPages.
- TCP/UDP: Choose to either filter TCP or UDP, or choose to filter both.
- Remark: Remark field, you can write comments by yourself.

IP Filtering

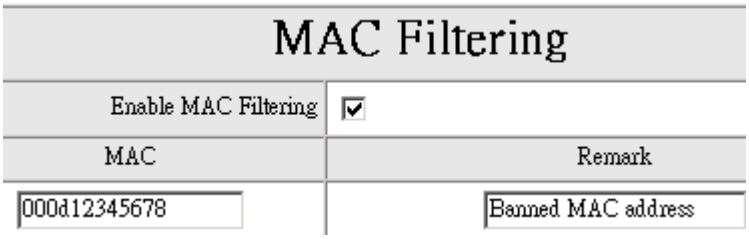
IP Filtering is to limit internal users from accessing the Internet.



- IP: Input the IP address that you want to filter; the limited IP address will be unable to transmit the data to the Internet.
- TCP/UDP: Choose to either filter TCP or UDP, or choose to filter both.
- Remark: Remark filed, you can write comments by yourself.

MAC Filtering

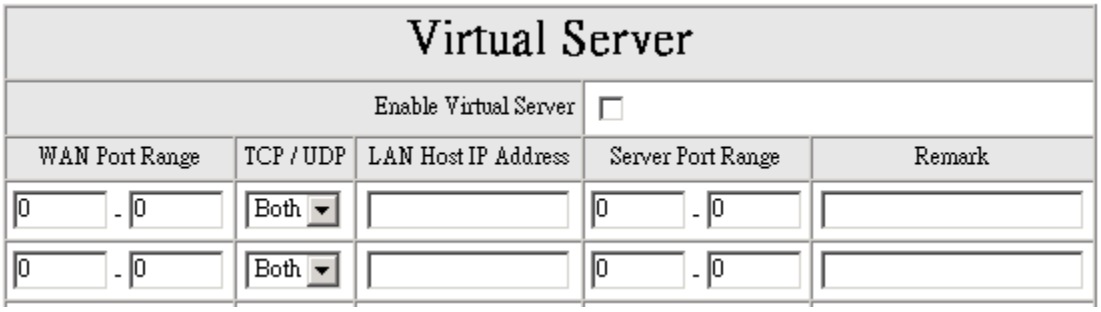
MAC (Media Access Control) address filtering is to filter the transmission of data by network card physical address.



MAC: input MAC that will limit accessing Internet PC.

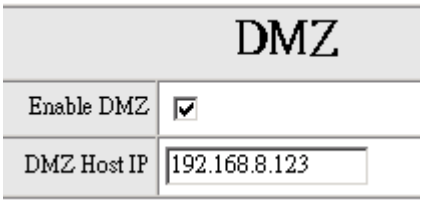
Virtual Server

Enabling the users on Internet to access the WWW, FTP and other services under your NAT. When remote user are accessing Web or FTP servers through WAN end IP address, it will be routed to the server at the internal LAN end and be routed to the server at the internal LAN end as appropriate in accordance with the externally required services



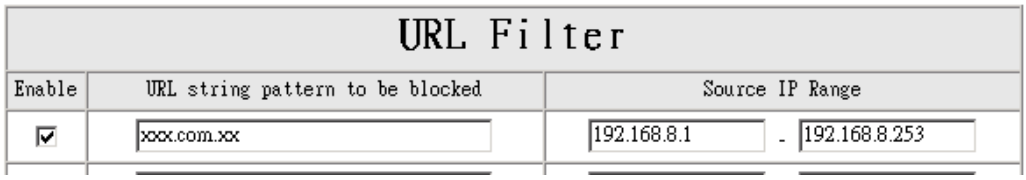
- WAN Port Range: Input the port on the WAN side.
- TCP/UDP: Select the communication protocols used by the server—TCP or UDP.
- LAN Host IP Address: Input IP address that provides various services servers.
- Server Port Range: Input the port used by the LAN host.

DMZ



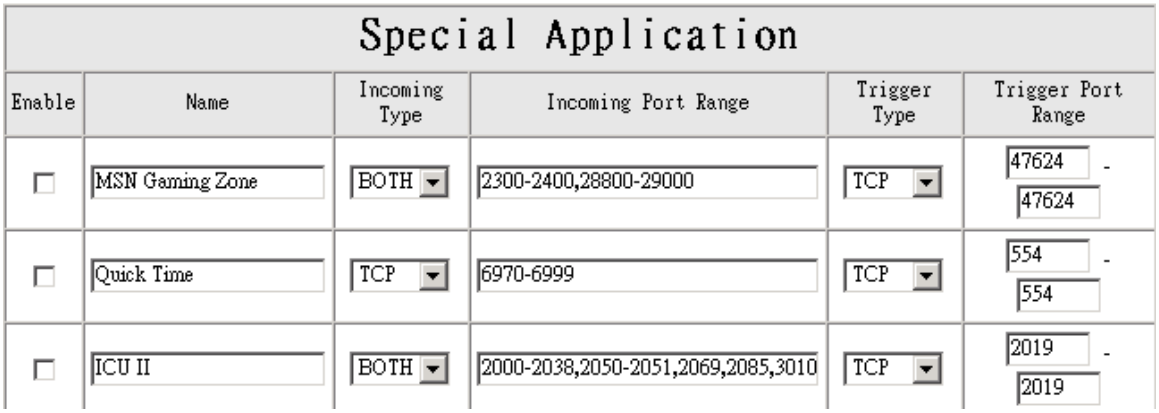
Lets the server on the LAN to be directly exposed to the Internet for accessing data. Either this function or the virtual server can be selected for use.

URL Filter※



URL filter is used to deny device from LAN accessing specific web sites. The system will block the URL that contains the string.

Special Applications※



Provide multiple connections for special applications.

- Name: The name of the special application.
- Incoming Type: The protocol used to trigger the special application.
- Incoming Port range: Port range on the WAN side that will be used to access the application.
- Trigger Type: The protocol used to trigger the application.
- Trigger Port Range: Port range used to trigger the application.

DoS Prevention Settings※

DoS Prevention Settings		
Enable DoS Prevention	☑	
Enable DoS Prevention on LAN	☑	
Whole System Flood	☑ SYN	50 (Packets/Second) [50 - 500]
	☑ FIN	50 (Packets/Second) [50 - 500]
	☐ UDP	100 (Packets/Second)
	☑ ICMP	50 (Packets/Second) [50 - 500]
Per-Source IP Flood	☑ SYN	30 (Packets/Second) [30 - 300]
	☑ FIN	30 (Packets/Second) [30 - 300]
	☐ UDP	100 (Packets/Second)

- Enable DoS Prevention: To prevent DoS from WAN.
- Enable DoS Prevention on LAN: To prevent DoS from LAN.
- Packet/Second: If the same packet type is more than the setting in one second, then it will be attacked. And VoIP gateway will block the IP if you checked “Enable Source IP Blocking”.
- Enable Source IP Blocking: Block the IP.
- Blocking Time: The time to block the IP.

Notice

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These functions are only supported for the second hardware. The second hardware can support more function, such as PPTP, LAN Interface Mode, LAN QoS, etc. There is an easiest way to make out the first and second hardware, please check Network Settings of User Interface. If there is LAN Interface Mode or PPTP in Network Settings, this model is the second hardware.

General

"IMPORTANT SAFETY INSTRUCTIONS - When using your telephone equipment, basic safety precautions should always be followed to reduce the risk of fire, electric shock and injury to persons, including the following:
-Do not use this product near water for example, near a bathtub, washbowl, and kitchen sink or laundry tub, in a wet basement or near a swimming pool.
-Avoid using a telephone (other than a cordless type) during an electrical storm. There may be a remote risk of electric shock from lightning.
-Do not use the telephone to report a gas leak in the vicinity of the leak.
-Use only the power cord and batteries indicated in this manual. Do not dispose of batteries in a fire. They may explode. Check with local codes for possible special disposal instructions.
SAVE THESE INSTRUCTIONS"

Telephone line cord

"CAUTION: To reduce the risk of fire, use only No. 26 AWG or larger UL Listed or CSA Certified Telecommunication Line Cord"

END OF THIS DOCUMENT